POLYCOM®

Polycom RMX™ 2000 System Administrator's Guide

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PATENT PENDING

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United States Federal Communication Commission (FCC)

Part 15: Class A Statement. This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. Test limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio-frequency energy and, if not installed and used in accordance with the instruction manuals, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his or her own expense.

Part 68: Network Registration Number. This equipment is registered with the FCC in accordance with Part 68 of the FCC Rules. This equipment is identified by the FCC registration number.

If requested, the FCC registration Number and REN must be provided to the telephone company.

Any repairs to this equipment must be carried out by Polycom Inc. or our designated agent. This stipulation is required by the FCC and applies during and after the warranty period.

United States Safety Construction Details:

- All connections are indoor only.
- Unit is intended for RESTRICTED ACCESS LOCATION.
- Unit is to be installed in accordance with the National Electrical Code.
- The branch circuit overcurrent protection shall be rated 20 A for the AC system.
- This equipment has a maximum operating ambient of 40°C, the ambient temperature in the rack shall not exceed this temperature.

To eliminate the risk of battery explosion, the battery should not be replaced by an incorrect type. Dispose of used batteries according to their instructions.

CE Mark R&TTE Directive

Polycom Inc., declares that the Polycom RMX™ 2000 is in conformity with the following relevant harmonized standards:

EN 60950-1:2001

EN 55022: 1998+A1:2000+A2:2003 class A

EN 300 386 V1.3.3: 2005

Following the provisions of the Council Directive 1999/CE on radio and telecommunication terminal equipment and the recognition of its conformity.

Canadian Department of Communications

This Class [A] digital apparatus complies with Canadian ICES-003.

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets telecommunication network protective, operational and safety requirements as prescribed in the appropriate Terminal Equipment Technical Requirements document(s). The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations. Repairs to certified equipment malfunctions, may give the telecommunications company causes to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

Regulatory Notices

Chinese Communication Certificate

声　明

此为 A 级产品，在生活环境中，该产品可能会造成无线电干扰。在这种情况下，可能需要用户对其干扰采取切实可行的措施。

Singapore Certificate

RMX 2000 complies with IDA standards G0916-07

Table of Contents

Conference Profiles 1-1
Conferencing Modes 1-3
Standard Conferencing 1-3
Supplemental Conferencing Features 1-5
Viewing Profiles 1-6
Profile Toolbar 1-7
Defining Profiles 1-8
Modifying an Existing Profile 1-23
Deleting a Conference Profile 1-24
Additional Conferencing Information 2-1
Video Session Modes 2-1
Dynamic Continuous Presence (CP) Mode 2-1
High Definition Video Switching Mode 2-2
Continuous Presence (CP) Conferencing 2-2
Video Resolutions in CP 2-3
Additional Video Resolutions in MPM+ Mode 2-4
Additional Intermediate Video Resolutions 2-4
Video Display with CIF, SD and HD Video Connections 2-5
Setting the Maximum CP Resolution for Conferencing ... 2-5
CP Conferencing with H.263 4CIF 2-6
H.263 4CIF Guidelines 2-7
High Definition Video Switching 2-8
HD VSW Guidelines 2-8
Enabling HD Video Switching 2-9
Modifying the HD Video Switching Threshold Bit Rate .. 2-9
Creating a High Definition Video Switching Profile 2-9
Monitoring High Definition Video Switching Conferences ... 2-11
H.239 2-12
Content Transmission Modes 2-12
Content Protocol 2-13
Defining Content Sharing Parameters for a Conference 2-14
Sending Content to Legacy Endpoints 2-16

Guidelines for Sending Content to Legacy Endpoints 2-16
Interoperability with Polycom CMA and DMA 2-17
Content Display on Legacy Endpoints 2-18
Enabling the Send Content to Legacy Endpoints Option 2-19
Changing the Default Layout for Displaying Content on Legacy Endpoints 2-20
Stopping a Content Session 2-22
Lecture Mode 2-23
Enabling Lecture Mode 2-23
Enabling the Automatic Switching 2-23
Selecting the Conference Lecturer 2-24
Lecture Mode Monitoring 2-25
Closed Captions 2-28
Enabling Closed Captions 2-29
Media Encryption 2-30
Encryption Flag Settings 2-33
Enabling Encryption in the Profile 2-33
Enabling Encryption at the Participant Level (IP Only) 2-34
Monitoring the Encryption Status 2-35
LPR – Lost Packet Recovery 2-36
Packet Loss 2-36
Causes of Packet Loss 2-36
Effects of Packet Loss on Conferences 2-36
Lost Packet Recovery 2-36
Lost Packet Recovery Guidelines 2-37
Enabling Lost Packet Recovery 2-37
Monitoring Lost Packet Recovery 2-38
Telepresence Mode 2-40
RMX 2000 Telepresence Mode Guidelines 2-40
System Level 2-40
Conference Level 2-41
Room (Participant/Endpoint) Level 2-41
RPX and TPX Video Layouts 2-42
Enabling Telepresence 2-45
Conference Level 2-45
Room (Participant/Endpoint) Level 2-46
Saving an Ongoing Conference as a Template 2-47

Starting an Ongoing Conference From a Template 2-48
Cascading Conferences - Star Topology 2-49
Enabling Cascading 2-51
Creating the Cascade-enabled Entry Queue 2-51
Creating the Dial-out Cascaded Link 2-53
Enabling Cascaded Conferences without Password 2-56
Monitoring Cascaded Conferences 2-57
Creating the Dial-out Link from a Conference Running on the MGC to the Conference Running on the RMX 2-58
Cascading Conferences - H.239-enabled MIH Topology 2-59
MIH Cascading Levels 2-59
MIH Cascading Guidelines 2-60
Master and Slave Conferences 2-60
Video Session Mode, Line Rate and Video Settings 2-61
H.239 Content Sharing 2-62
Setting up MIH Cascading Conferences 2-63
RMX to RMX Cascading 2-63
MGC to RMX 2000 Cascading 2-71
Starting and Monitoring MIH Cascading Conferences 2-81
Monitoring Participants in an MIH Cascaded Conference 2-81
Viewing Participant Properties 2-83

Meeting Rooms 3-1

Meeting Rooms List 3-2
Meeting Room Toolbar & Right-click Menu 3-4
Creating a New Meeting Room 3-5

Entry Queues, Ad Hoc Conferences and SIP Factories 4-1

Entry Queues 4-1
Defining a New Entry Queue 4-3
Listing Entry Queues 4-7
Modifying the EQ Properties 4-8
Transit Entry Queue 4-8
Setting a Transit Entry Queue 4-8
Ad Hoc Conferencing 4-10
Gateway to Polycom® Distributed Media Application™ (DMA™) 7000 4-11
SIP Factories 4-12

Creating SIP Factories 4-12
Address Book 5-1
Viewing the Address Book 5-2
Displaying and Hiding the Address Book 5-2
Adding a Participant to the Address Book 5-4
Adding a new participant to the Address Book Directly 5-4
Adding a Participant from an Ongoing Conference to the Address Book 5-12
Modifying Participants in the Address Book 5-13
Deleting Participants from the Address Book 5-14
Searching the Address Book 5-14
Filtering the Address Book 5-15
Participant Groups 5-17
Adding a New Group to the Address Book 5-17
Deleting a Group from the Address Book 5-18
Modifying a Group in the Address Book 5-19
Importing and Exporting Address Books 5-20
Exporting an Address Book 5-20
Importing an Address Book 5-21
Integrating the Polycom CMA™ Address Book with the RMX 5-22
Reservations 6-1
Guidelines 6-1
System 6-1
Resources 6-1
Reservations 6-3
Using the Reservation Calendar 6-4
Toolbar Buttons 6-4
Reservations Views 6-5
Week View 6-6
Day View 6-6
Today View 6-6
List View 6-7
Changing the Calendar View 6-8
Scheduling Conferences Using the Reservation Calendar 6-10
Creating a New Reservation 6-10
Managing Reservations 6-18
Guidelines 6-18

Viewing and Modifying Reservations 6-18
Using the Week and Day views of the Reservations Calendar 6-18
Deleting Reservations 6-20
Searching for Reservations using Quick Search 6-21
Operator Assistance & Participant Move 7-1
Operator Conferences 7-1
Defining the Components Enabling Operator Assistance 7-3
Defining a Conference IVR Service with Operator Assistance Options 7-4
Defining an Entry Queue IVR Service with Operator Assistance Options 7-7
Defining a Conference Profile for an Operator Conference 7-9
Defining an Ongoing Operator Conference 7-16
Saving an Operator Conference to a Template 7-21
Starting an Operator Conference from a Template 7-22
Monitoring Operator Conferences and Participants Requiring Assistance 7-23
Requesting Help 7-23
Participant Alerts List 7-25
Moving Participants Between Conferences 7-25
Moving Participants 7-27
Conference Templates 8-1
Guidelines 8-1
Using Conference Templates 8-3
Toolbar Buttons 8-4
Creating a New Conference Template 8-5
Creating a new Conference Template from Scratch 8-5
Saving an Ongoing Conference as a Template 8-12
Saving an Operator Conference to a Template 8-12
Starting an Ongoing Conference From a Template 8-14
Starting an Operator Conference from a Template 8-15
Scheduling a Reservation From a Conference Template 8-16
Deleting a Conference Template 8-18
Conference and Participant Monitoring 9-1
General Monitoring 9-2

Conference Level Monitoring 9-3
Monitoring Operator Conferences and Participants Requiring Assistance 9-11
Requesting Help 9-11
Participant Alerts List 9-13
Participant Level Monitoring 9-14
IP Participant Properties 9-15
Monitoring ISDN/PSTN Participants 9-25
Recording Conferences 10-1
Configuring the RMX to enable Recording 10-1
Defining the Recording Link 10-1
Enabling the Recording Features in a Conference IVR Service 10-3
Enabling the Recording in the Conference Profile 10-4
Managing the Recording Process 10-6
Using the RMX Web Client to Manage the Recording Process 10-6
Using DTMF Codes to Manage the Recording Process 10-8
Conference Recording with Codian IP VCR 10-9
Users, Connections and Notes 11-1
Listing Users 11-2
Adding a New User 11-3
Deleting a User 11-4
Changing a User's Password 11-5
Connections 11-6
Viewing the Connections List 11-6
Notes 11-7
Using Notes 11-7
Network Services 12-1
IP Network Services 12-2
Management Network (Primary) 12-2
Default IP Service (Conferencing Service) 12-2
Modifying the Management Network 12-3
Modifying the Default IP Network Service 12-9
IP Network Monitoring 12-23
ISDN/PSTN Network Services 12-27
Adding/Modifying ISDN/PSTN Network Services 12-28

Obtaining ISDN/PSTN required information 12-28
Modifying an ISDN/PSTN Network Service 12-36
IVR Services 13-1
IVR Services List 13-2
IVR Services Toolbar 13-3
Adding Languages 13-4
Defining a New Conference IVR Service 13-9
Defining a New Conference IVR Service 13-9
Entry Queues IVR Service 13-26
Defining a New Entry Queue IVR Service 13-26
Setting a Conference IVR Service or Entry Queue IVR Service as the Default Service 13-31
Modifying the Conference or Entry Queue IVR Service Properties 13-32
Replacing the Music File 13-33
Adding a Music File 13-33
Creating Audio Prompts and Video Slides 13-35
Recording an Audio Message 13-35
Creating a Welcome Video Slide 13-39
Default IVR Prompts and Messages 13-40
Volume Control of IVR Messages, Music and Roll Call 13-44
The Call Detail Record (CDR) Utility 14-1
The CDR File 14-2
CDR File Formats 14-2
CDR File Contents 14-3
Viewing, Retrieving and Archiving Conference Information 14-5
Viewing the Conference Records 14-5
Refreshing the CDR List 14-6
Retrieving and Archiving Conference CDR Records 14-7
Gateway Calls 15-1
Call Flows 15-1
Direct Dialing 15-1
Gateway IVR 15-6
Interoperability with CMA 15-9
Connection Indications 15-9
Gateway Functionality 15-10
Configuring the Gateway Components on the RMX 15-12

Defining the IVR Service for Gateway Calls 15-12
Defining the Conference Profile for Gateway Calls 15-16
Defining the Gateway Profile 15-17
Displaying the Connection Information - System Configuration 15-21
Monitoring Ongoing Gateway Sessions 15-22
Gateway Session Parameters 15-22
Connected Participant Parameters 15-23
Dialing to Polycom® DMA™ 7000 15-24
Direct Dialing from ISDN/PSTN Endpoint to IP Endpoint via a Meeting Room 15-25
RMX Administration and Utilities 16-1
RMX Manager 16-1
Installing RMX Manager 16-1
Running RMX Manager 16-4
System and Participant Alerts 16-6
System Alerts 16-7
Participant Alerts 16-9
System Configuration 16-10
Modifying System Flags 16-10
LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values 16-17
Manually Adding and Deleting System Flags 16-19
Auto Layout Configuration 16-30
Customizing the Default Auto Layout 16-30
RMX Time 16-33
Altering the clock 16-33
Resource Management 16-35
Resource Capacity 16-35
Resource Capacity Modes 16-36
Resource Usage 16-37
Video/Voice Port Configuration 16-39
Flexible Resource Capacity Mode 16-39
Fixed Resource Capacity 16-40
Configuring the Video/Voice Resources in MPM Mode 16-41

Configuring the Video/Voice Resources in MPM+ Mode 16-42
Flexible Resource Capacity 16-42
Forcing Video Resource Allocation to CIF Resolution . 16-46
Resource Report 16-48
Displaying the Resource Report 16-48
Resource Report Display in Flexible Resource Capacity Mode™ 16-49
Resource Report in Fixed Resource Capacity Mode™ . 16-51
ISDN/PSTN 16-52
Port Usage 16-53
Setting the Port Usage Threshold 16-53
Port Usage Gauges 16-54
Port Gauges in Flexible/Fixed Capacity Modes 16-55
System Information 16-56
SNMP (Simple Network Management Protocol) 16-59
Detailed Description 16-59
MIB (Management Information Base) Files 16-59
Private MIBS 16-59
Support for MIB-II Sections 16-60
The Alarm-MIB 16-60
H.341-MIB (H.341 – H.323) 16-60
Standard MIBs 16-60
Traps 16-62
Status Trap Content 16-63
Defining the SNMP Parameters in the RMX 16-64
Multilingual Setting 16-70
Customizing the Multilingual Setting 16-70
Software Management 16-71
Using Software Management 16-71
Notification Settings 16-73
Logger Diagnostic Files 16-75
Auditor 16-78
Auditor Files 16-78
Auditor Event History File Storage 16-78
Retrieving Auditor Files 16-79
Auditor File Viewer 16-81

Audit Events 16-84
Alerts and Faults 16-84
Transactions 16-85
ActiveX Bypass 16-88
Installing ActiveX 16-88
Resetting the RMX 16-90
Hardware Monitoring 17-1
Viewing the Status of the Hardware Components 17-1
HW Monitor Pane Toolbar 17-3
Viewing Hardware Component's Properties 17-4
Diagnostic Mode 17-13
Performing Diagnostics 17-13
Diagnostics Monitoring 17-16
MCU Monitor 17-16
Cards Monitor 17-17
Error Buffer 17-18
Appendix A - Disconnection Causes A-1
IP Disconnection Causes. A-1
ISDN Disconnection Causes A-10
Appendix B - Alarms and Faults B-1
Alarms B-1
Appendix C - CDR Fields - Unformatted File C-1
The Conference Summary Record C-3
Event Records C-5
Standard Event Record Fields C-5
Event Types C-6
Event Specific Fields C-16
Disconnection Cause Values C-55
MGC Manager Events that are not Supported by the RMX 2000 ..... C-59
Appendix D - Ad Hoc Conferencing and External Database Authentication D-1
Ad Hoc Conferencing without Authentication D-2
Ad Hoc Conferencing with Authentication D-3
Entry Queue Level - Conference Initiation Validation with an External Database Application D-4
Conference Access with External Database Authentication D-6
Conference Access Validation - All Participants (Always) D-7

Conference Access Validation - Chairperson Only (Upon Request) D-9
System Settings for Ad Hoc Conferencing and External Database Authentication D-11
Ad Hoc Settings D-11
Authentication Settings D-11
MCU Configuration to Communicate with an External Database Application D-13
Enabling External Database Validation for Starting New Ongoing Conferences D-15
Enabling External Database Validation for Conferences Access D-16
Appendix E - Participant Properties Advanced Channel Information E-1
Appendix F - Secure Communication Mode F-1
Switching to Secure Mode F-1
Purchasing a Certificate F-1
Installing the Certificate F-3
Creating/Modifying System Flags F-5
Enabling Secure Communication Mode F-5
Alternate Management Network F-6
Securing an External Database F-7
Appendix G - Configuring Direct Connections to RMX . G-1
Management Network (Primary) G-1
Alternate Management Network G-1
Configuring the Workstation G-2
Connecting to the Management Network G-6
Connecting to the Alternate Management Network G-8
Connecting to the RMX via Modem G-9
Procedure 1: Install the RMX Manager G-9
Procedure 2: Configure the Modem G-9
Procedure 3: Create a Dial-up Connection G-10
Procedure 4: Connect to the RMX G-15
Appendix H - Setting the RMX for Integration Into Microsoft OCS Environment H-1
Configuring the OCS for RMX 2000 ..... H-2

Setting the Trusted Host and Static Route for RMX in the OCS H-2
Creating the Security (TLS) Certificate in the OCS and Exporting the Certificate to the RMX Workstation H-7
Retrieving the Certificate from the OCS to the RMX Workstation H-13
Optional. Creating the Certificate Password File (certPassword.txt) H-16
Optional. Setting the Static Route & Trusted Host for RMX in the Load Balancer Server H-17
Configuring the RMX 2000 for Microsoft OCS 2007 Integration .. H-19
Modify the RMX Management Network Service to Include the DNS Server H-19
Defining a SIP Network Service in the RMX H-21
Polycom RMX System Flag Configuration H-25
Dialing to an Entry Queue, Meeting Room or Conference ... H-28
Active Alarms and Troubleshooting H-29
Active Alarms H-29
Troubleshooting H-31
Known Issues H-31

1

Conference Profiles

Profiles stored on the MCU enable you to define all types of conferences. Profiles include conference parameters such as Bit Rate, Video Layout, Encryption, etc.

A maximum of 40 *Conference Profiles* can be defined.

Conference Profiles are saved to *Conference Templates* along with all participant parameters, including their *Personal Layout* and *Video Forcing* settings, enabling administrators and operators to create, save, schedule and activate identical conferences. For more information see Chapter 8, "*Conference Templates*" .

The RMX is shipped with a default *Conference Profile* which allows users to immediately start standard ongoing conferences. Its settings are as follows:

Table 1-1 *Default Conference Profile Settings*

Setting	Value
Profile Name	Factory Video Profile
Bit Rate	384Kbps
H.239 Settings	Graphics
High Definition Video Switching	Disabled
Operator Conference	Disabled
Encryption	Disabled
LPR	Enabled for CP Conferences
Auto Terminate	• After last participant quits - Enabled • When last participant remains - Disabled

Table 1-1 *Default Conference Profile Settings (Continued)*

Setting	Value
Echo Suppression	Enabled
Keyboard Noise Suppression	Disabled
Video Quality	Sharpness
Video Clarity™	Enabled
Content Video Definition	• Content Settings: Graphics • Content Protocol: Up to H.264
Send Content to Legacy Endpoints	Enabled
Layout	Auto Layout - Disabled Same Layout - Disabled
Skin	Polycom
IVR Name	Conference IVR Service

This Profile is automatically assigned to the following conferencing entities:

Name	ID
Meeting Rooms	
Maple_Room	*1001*
Oak_Room	*1002*
Juniper_Room	*1003*
Fig_Room	*1004*
Entry Queue	
Default EQ	*1000*

Conferencing Modes

Standard Conferencing

When defining a new video Profile, you select the parameters that determine the video display on the participant's endpoint and the quality of the video. When defining a new conference Profile, the system uses default values for standard conferencing. Standard conferencing enable several participants to be viewed simultaneously and each connected endpoint uses its highest video, audio and data capabilities up to the maximum bit rate set for the conference.

The main parameters that define the quality of a video conference are:

- **Bit Rate -** The transfer rate of video and audio streams. The higher the bit rate, the better the video quality.
- **Audio Algorithm -** The audio compression algorithm determines the quality of the conference audio.
- **Video protocol, video format, frame rate, annexes, and interlaced video mode** - These parameters define the quality of the video images. The RMX will send video at the best possible resolution supported by endpoints regardless of the resolution received from the endpoints.
  - — When Sharpness is selected as the Video Quality setting in the conference Profile, the RMX will send 4CIF (H.263) at 15fps instead of CIF (H.264) at 30fps.
  - — H.264 protocol provides better compression of video images in bit rates lower than 384 Kbps and it will be automatically selected for the endpoint if it supports H.264.
  - — When working with RMXs at low bit rates (128, 256, or 384Kbps), HDX endpoints will transmit SD15 resolution instead of 2CIF resolution.

  When using 1x1 conference layout, the RMX transmits the same resolution it receives from the endpoint.
- **Lost Packet Recovery (LPR) -** LPR creates additional packets that contain recovery information used to reconstruct packets that are lost during transmission.

- **Video Clarity -** Video Clarity feature applies video enhancing algorithms to incoming video streams of resolutions up to and including SD.
- **Supported resolutions:**
  - **H.261 CIF/QCIF –** Is supported in Continuous Presence (CP) conferences at resolutions of 288 x 352 pixels (CIF) and 144 x 176 pixels (QCIF). Both resolutions are supported at frame rates of up to 30 frames per second.
  - **H.263 4CIF** - A high video resolution available to H.263 endpoints that are not H.264 enabled. It is only supported for conferences in which the video quality is set to sharpness and for lines rates of 384kbps to 1920kbps.
  - **Standard Definition (SD)** - A high quality video protocol which uses the H.264 video algorithm. It enables HD compliant endpoints to connect to Continuous Presence conferences at resolutions of 720X576 pixels for PAL systems and 720X480 pixels for NTSC systems. Bit rates for SD range from 256Kbps to 2Mbps. For more information, see "*Video Resolutions in CP*" on page **2-3**.
  - **High Definition (HD) –** HD is an ultra-high quality video resolution. Depending on the RMX's Card Configuration mode, compliant endpoints are able to connect to conferences at resolutions ranging from 720p (1280 x 720 pixels) to 1080p (1920 x 1080 pixels) (in MPM+ Mode) at bit rates ranging from 1024 Kbps to 4 Mbps (6 Mbps with HD VSW). For more information, see "*Video Resolutions in CP*" on page **2-3**.
  - **Operator Conferences** - Offers additional conference management capabilities to the RMX users, enabling them to attend to participants with special requirements and acquire participant details for billing and statistics. This service is designed usually for large conferences that require the personal touch. Operator assistance is available in both MPM and MPM+ *Card Configuration Modes*. For more information, see Chapter 7, "*Operator Assistance & Participant Move*" on page **7-1**.

Supplemental Conferencing Features

In addition to *Standard Conferencing* the following features can be enabled:

- **H.239 –** Allows compliant endpoints to transmit and receive two simultaneous streams of conference data to enable Content sharing. H.239 is also supported in cascading conferences. Both H.263 and H.264 Content sharing protocols are supported. If all endpoints connected to the conference have H.264 capability, Content is shared using H.264, otherwise Content is shared using H.263.

  For more information, see "*H.239*" on page **2-12**.

- **Lecture Mode –** The lecturer is seen by all participants in full screen while the lecturer views all conference participants in the selected video layout.

  For more information, see "*Lecture Mode*" on page **2-23**.

- **Presentation Mode –** When the current speaker's speech exceeds a predefined time (30 seconds), the conference layout automatically changes to full screen, displaying the current speaker as the conference lecturer on all the participants' endpoints. During this time the speaker's endpoint displays the previous conference layout. When another participant starts talking, the Presentation Mode is cancelled and the conference returns to its predefined video layout. Presentation mode is available with *Auto Layout* and *Same Layout*.
  - — If the speaker in a video conference is an Audio Only participant, the Presentation Mode is disabled for that participant.
  - — Video forcing works in the same way as in Lecture Mode when Presentation Mode is activated, that is, forcing is only enabled at the conference level, and it only applies to the video layout viewed by the lecturer.

- **Telepresence Mode** - enables the connection of numerous high definition telepresence rooms and of different models (such as TPX and RPX) into one conference maintaining the telepresence experience. This mode is enabled by a special license.

- **Encryption –** Used to enhance media security at conference and participant levels. For more information, see "*Media Encryption*" on page **2-30**.

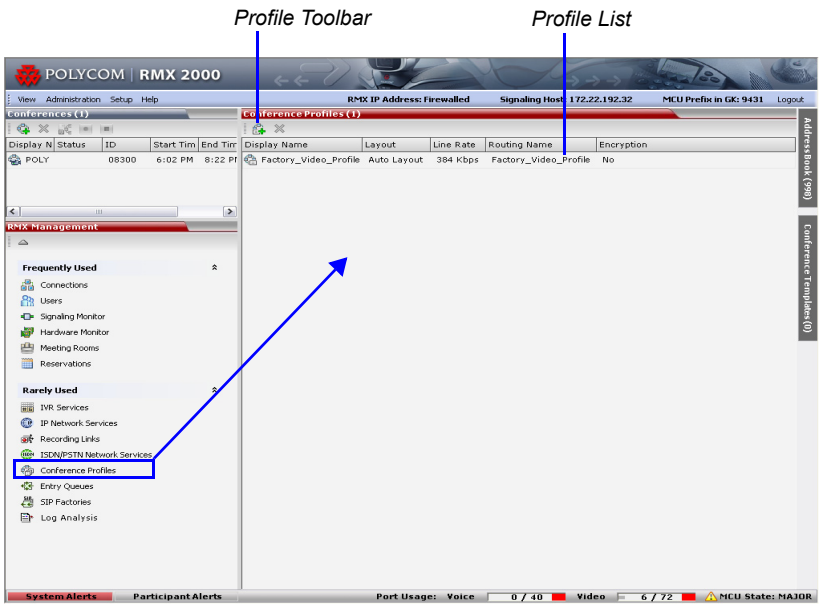
- **Conference Recording** - The RMX enables audio and video recording of conferences using Polycom RSS 2000 recording system.

Viewing Profiles

Conference Profiles are listed in the *Conference Profiles* list pane.

To list Conference Profiles:

1. In the *RMX Management* pane, expand the *Rarely Used* list.
2. Click the **Conference Profiles** button.
   The *Conference Profiles* are displayed in the *List* pane.



The following *Conference Profile* properties are displayed in the *List* pane:

Table 1-2 *Conference Profiles Pane Columns*

Field	Description
Name	The name of the *Conference Profile*.
Layout	Displays either "*Auto Layout*" or an icon of the layout selected for the profile. For information about video layouts, see Table 1-9 "*Video Layout Options*" on page **1-19**.

Table 1-2 Conference Profiles Pane Columns (Continued)

Field	Description
Line Rate	The maximum bit rate at which endpoints can connect to the conference.
Routing Name	Displays the Routing Name defined by the user or automatically generated by the system.
Encryption	Displays if media encryption is enabled for the Profile (Yes). For more information about encryption, see "*Media Encryption"* on page 2-30.

Profile Toolbar

The Profile toolbar provides quick access to the Profile functions:

Table 1-3 Profile Toolbar buttons

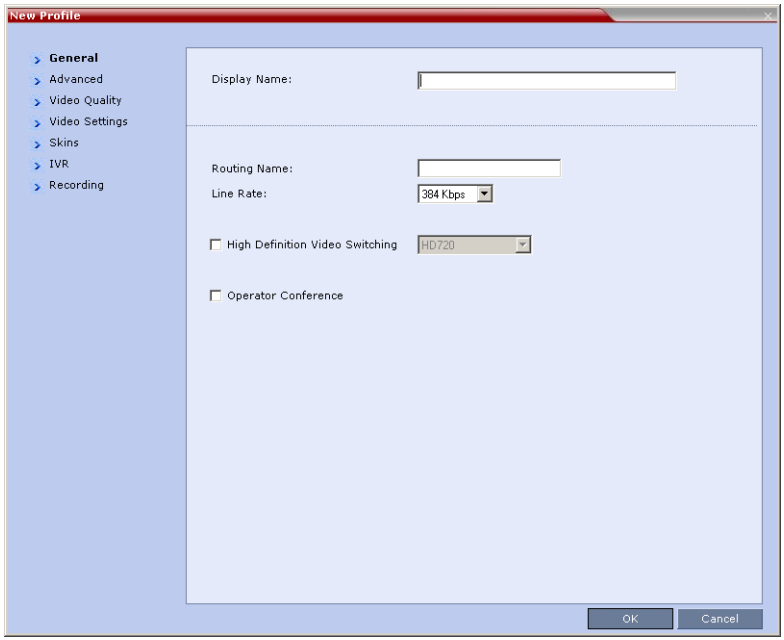
Button	Button Name	Descriptions
	New Profile	To create a new Profile.
	Delete Profile	To delete a profile, click the Profile name and then click this button.

Defining Profiles

Profiles are the basis for the definition of all ongoing conferences, Reservations, Meeting Rooms, Entry Queues, and Conference Templates and they contain only conference properties.

To define a new Profile:

1. In the *RMX Management* pane, click **Conference Profiles**.
2. In the *Conference Profiles* pane, click the **New Profile** button. The *New Profile – General* dialog box opens.



The RMX displays the default settings, so you need only define the Profile name.

3 Define the Profile name and, if required, the Profile general parameters:

Table 1-4 *New Profile - General Parameters*

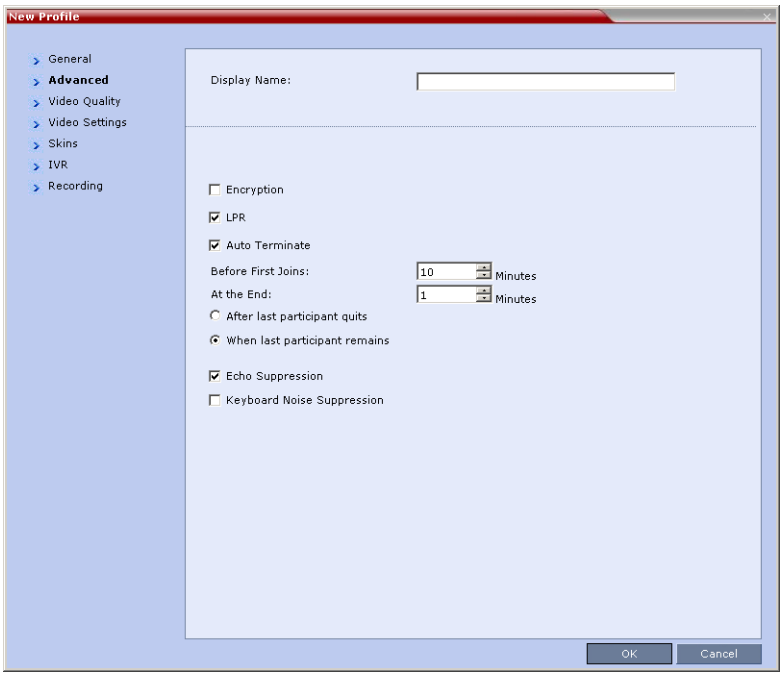
Field/Option	Description
Display Name	Enter a unique Profile name, as follows: • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. It is recommended to use a name that indicates the Profile type, such as Operator conference or Video Switching conference. **Note:** This is the only parameter that must be defined when creating a new profile.
Routing Name	Enter the Profile name using ASCII characters set. The Routing Name can be defined by the user or automatically generated by the system if no Routing Name is entered as follows: • If an all ASCII text is entered in Display Name, it is used also as the Routing Name. • If any combination of Unicode and ASCII text (or full Unicode text) is entered in Display Name, the ID (such as Conference ID) is used as the Routing Name.
Line Rate	Select the conference bit rate. The line rate represents the combined video, audio and Content rate. The default setting is 384 Kbps.

Table 1-4 *New Profile - General Parameters (Continued)*

Field/Option	Description
High Definition Video Switching	If the *Operator Conference* option is selected, this option is disabled, and the selection is cleared. When selected, the conference is ultra-high quality video resolution, in a special conferencing mode which implies that all participants must connect at the same line rate and use HD video and all participants with endpoints not supporting HD will connect as secondary (audio only). For more information, see "*High Definition Video Switching"* on page **2-8**. Select the High Definition resolution; select either HD 720p or HD 1080p (in MPM+ mode only). If HD 1080p is selected, endpoints that do not support HD 1080p resolution are connected as Secondary (Audio Only) participants. **Notes:** • High Definition Video Switching conferencing mode is unavailable to ISDN participants.
Operator Conference	Select this option to define the profile of an Operator conference. An Operator conference can only be a Continuous Presence conference, therefore when selected, the *High Definition Video Switching* option is disabled and cleared. When defining an *Operator Conference*, the *Send Content to Legacy Endpoints* option in the *Video Settings* tab is cleared and disabled. For more information, see Chapter 7, "*Operator Assistance & Participant Move"* on page **7-1**.

4 Click the **Advanced** tab.

The *New Profile – Advanced* dialog box opens.



5 Define the following parameters:

Table 1-5 *New Profile - Advanced Parameters*

Field/Option	Description
Encryption	Select this check box to activate encryption for the conference. For more information, see *RMX 2000 Administrator's Guide*, "*Media Encryption"* on page **2-30**.
LPR	When selected (default for CP conferences), Lost Packet Recovery creates additional packets that contain recovery information used to reconstruct packets that are lost during transmission. LPR is automatically disabled if High Definition Video Switching is selected. For more information, see *RMX 2000 Administrator's Guide, "LPR – Lost Packet Recovery"* on page **2-36**.

Table 1-5 *New Profile - Advanced Parameters (Continued)*

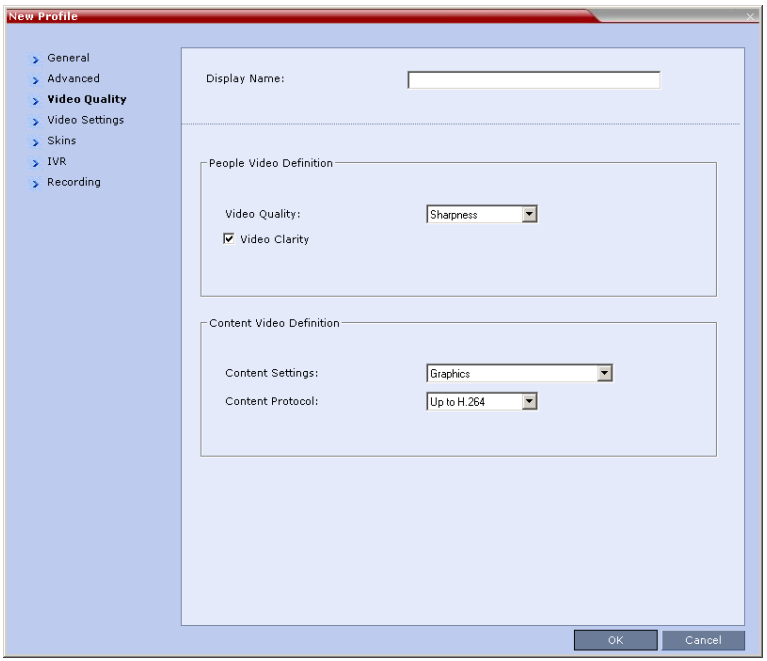
Field/Option	Description
Auto Terminate	When selected (default), the conference automatically ends when the termination conditions are met: ***Before First Joins*** — No participant has connected to a conference during the *n* minutes after it started. Default idle time is 10 minutes. ***At the End - After Last Quits*** — All the participants have disconnected from the conference and the conference is idle (empty) for the predefined time period. Default idle time is 1 minute. ***At the End - When Last Participant Remains*** — Only one participant is still connected to the conference for the predefined time period (excluding the recording link which is not considered a participant when this option is selected). This option should be selected when defining a Profile that will be used for Gateway Calls and you want to ensure that the call is automatically terminated when only one participant is connected. Default idle time is 1 minute. **Note:** The selection of this option is automatically cleared and disabled when the *Operator Conference* option is selected. The Operator conference cannot automatically end unless it is terminated by the RMX User.
Echo Suppression	When enabled (default), an algorithm is used to search for and detect echo outside the normal range of human speech (such as echo) and automatically mute them when detected. Clear this option to disable the Echo Suppression algorithm. **Notes:** • This option is activated only in *MPM+ Card Configuration Mode*. • The CMA uses the *Profiles* that are stored in the RMX. When the *Echo Suppression* is enabled, it will be enabled in the conference that is started from the CMA with that *Profile*. However, the CMA does not display an indication that this option is enabled for the conference.

Table 1-5 *New Profile - Advanced Parameters (Continued)*

Field/Option	Description
Keyboard Noise Suppression	Select this option to let the system use an algorithm to search for and detect keyboard noises and automatically mute them when detected. **Notes:** • This option is activated only in *MPM+ Card Configuration Mode*. • The CMA uses the *Profiles* that are stored in the RMX. When the *Keyboard Noise Suppression* is enabled, it will be enabled in the conference that is started from the CMA with that *Profile*. However, the CMA does not display an indication that this option is enabled for the conference.

6 Click the **Video Quality** tab.

The *New Profile – Video Quality* dialog box opens.



7 Define the following parameters:

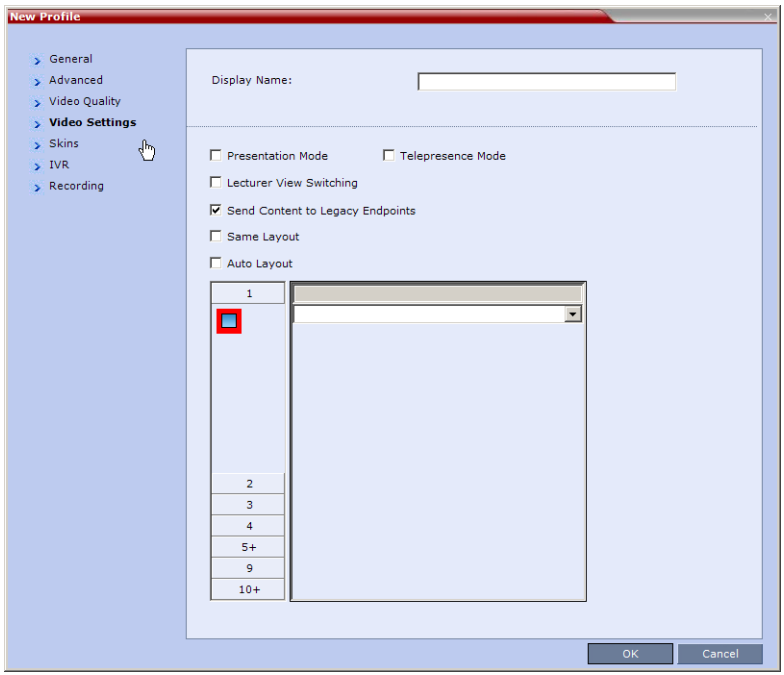
Table 1-6 *New Profile - Video Quality Parameters*

Field/Option	Description
People Video Definition	
Video Quality	Depending on the amount of movement contained in the conference video, select either: • ***Motion*** – for a higher frame rate without increased resolution. When selected, Video Clarity is disabled. • ***Sharpness*** – for higher video resolution and requires more system resources. **Note:** When Sharpness is selected as the Video Quality setting in the conference Profile, the RMX will send 4CIF (H.263) at 15fps instead of CIF (H.264) at 30fps. For more information, see *RMX 2000 Administrator's Guide*, "*Video Resolutions in CP"* on page **2-3**.
Video Clarity™	When enabled (default), *Video Clarity* applies video enhancing algorithms to incoming video streams of resolutions up to and including SD. Clearer images with sharper edges and higher contrast are sent back to all endpoints at the highest possible resolution supported by each endpoint. All layouts, including 1x1, are supported. Video Clarity is enabled only when *Video Quality* is set to *Sharpness* (default setting) and is disabled when *Video Quality* is set to *Motion*. Video Clarity can only be enabled for Continuous Presence conferences in MPM+ Card Configuration Mode.

Table 1-6 *New Profile - Video Quality Parameters (Continued)*

Field/Option	Description
Content Video Definition	
Content Settings	Select the transmission mode for the Content channel: • ***Graphics*** — basic mode, intended for normal graphics • ***Hi-res Graphics*** — a higher bit rate intended for high resolution graphic display • ***Live Video*** — Content channel displays live video Selection of a higher bit rate for the Content results in a lower bit rate for the people channel. For more information, see *RMX 2000 Administrator's Guide*, "*H.239"* on page **2-12**.
Content Protocol	**H.263** – Content is shared using *H.263* even if some endpoints have *H.264* capability. **Up to H.264** – *H.264* is the default Content sharing algorithm. When selected: • Content is shared using *H.264* if all endpoints have *H.264* capability. • Content is shared using *H.263* if all endpoints do not have *H.264* capability. • Endpoints that do not have at least *H.263* capability can connect to the conference but cannot share Content.

8 Click the **Video Settings** tab.
The *New Profile - Video Settings* dialog box opens.



9 Define the video display mode and layout using the following parameters:

Table 1-7 *Profile Properties - Video Settings*

Field/Option	Description
Presentation Mode	Select this option to activate the Presentation Mode. In this mode, when the current speaker speaks for a predefined time (30 seconds), the conference changes to Lecture Mode. When another participant starts talking, the Presentation Mode is cancelled and the conference returns to the previous video layout.

Table 1-7 *Profile Properties - Video Settings (Continued)*

Field/Option	Description
Lecture View Switching	Select this option to enable automatic switching of participants on the Lecturer's screen when Lecture Mode is enabled for the conference. The automatic switching is enabled when the number of participants exceeds the number of video windows displayed on the Lecturer's screen. **Note:** Lecture Mode is enabled in the *Conference Properties – Participants* tab. For more information, see "*Lecture Mode"* on page **2-23**.
Send Content to Legacy Endpoints	When enabled (default), Content can be sent to H.323/ SIP/ISDN endpoints that do not support H.239 Content (legacy endpoints) over the video (people) channel. For more details, see Chapter 2, "*Sending Content to Legacy Endpoints" on page* **2-16**. **Notes:** • This option is enabled only in **MPM+** *Card Configuration Mode* and *Resource Allocation Mode* is set to **Flexible Mode**. • This option is valid when sending Content as a separate stream is enabled in the *System Configuration* and the flag: ENABLE_H239 is set to YES. • If *High Definition Video Switching* option is selected in the *Conference Profile - General* tab, the *Send Content to Legacy Endpoints* selection is cleared and the option is disabled. • If the *Same Layout* option is selected, the *Send Content to Legacy Endpoints* selection is cleared and is disabled.
Same Layout	Select this option to force the selected layout on all participants in a conference. Displays the same video stream to all participants and personal selection of the video layout is disabled. In addition, if participants are forced to a video layout window, they can see themselves.

Table 1-7 *Profile Properties - Video Settings (Continued)*

Field/Option	Description
Auto Layout	Select this option to let the system automatically select the conference layout based on the number of participants currently connected to the conference. When a new video participant connects or disconnects, the conference layout automatically changes to reflect the new number of video participants. For more information, see Table 1-8 "*Auto Layout – Default Layouts"* on page **1-18**. The default Auto Layout settings can be customized by modifying default Auto Layout system flags in the System Configuration file. For more information see, "*Auto Layout Configuration"* on page **16-30**.
Telepresence Mode	Select this option to enable the Telepresence Mode in the Conference. **Notes:** • This field is enabled only if the RMX system is licensed for Telepresence Mode. • If the *Auto Layout* option is selected, the *Telepresence Mode* option is disabled.

Table 1-8 *Auto Layout – Default Layouts*

Number of Video Participants	Auto Layout Default Settings
0–2	
3	
4–5	
6–7	
8+	

The RMX supports the VUI addition to the H.264 protocol for endpoints that transmit wide video (16:9) in standard 4SIF resolution.

10 To select the *Video Layout* for the conference, click the required number of windows from the layouts bar and then select the windows array.

The selected layout appears in the *Video Layout* pane.

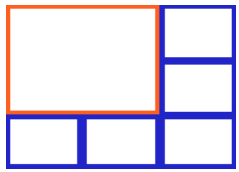
Table 1-9 *Video Layout Options*

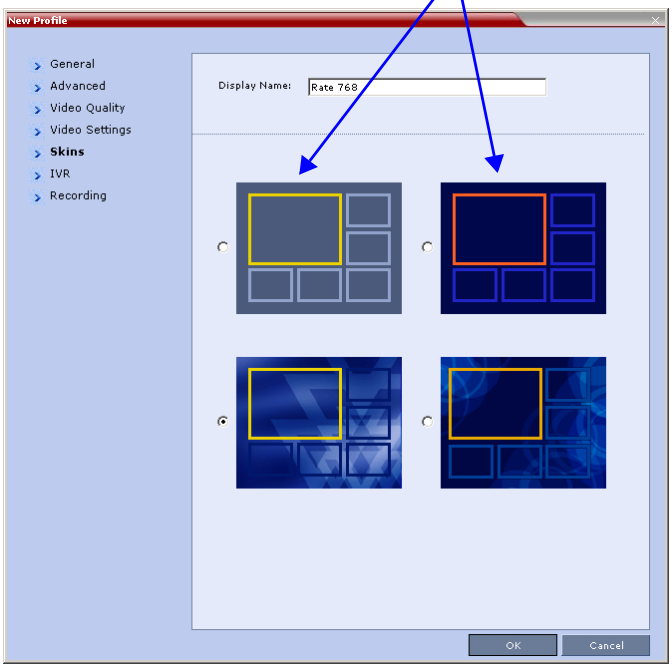
Number of Video Windows	Available Video Layouts
1	
2	
3	
4	
5+	
9	
10+	

When there is a change of speaker in a Continuous Presence conference, the transition is set by default to fade in the current speaker while fading out the previous speaker.
To make this transition visually pleasant, fading in the current speaker while fading out the previous speaker is done over a period of 500 milliseconds.
The *Fade In / Fade Out* feature can be disabled by adding a new flag to the *System Configuration*. The *Value* of the new flag must be: FADE_IN_FADE_OUT=NO.
For more information about *System Flags*, see the *RMX 2000 Administrator's Guide*, Chapter 16, "*System Configuration"* on page **16-10**.

11 Click the **Skins** tab to modify the background and frames. The *New Profile - Skins* dialog box opens.

In Classic View (for the first two skin options) the frames fill the screen with their borders touching



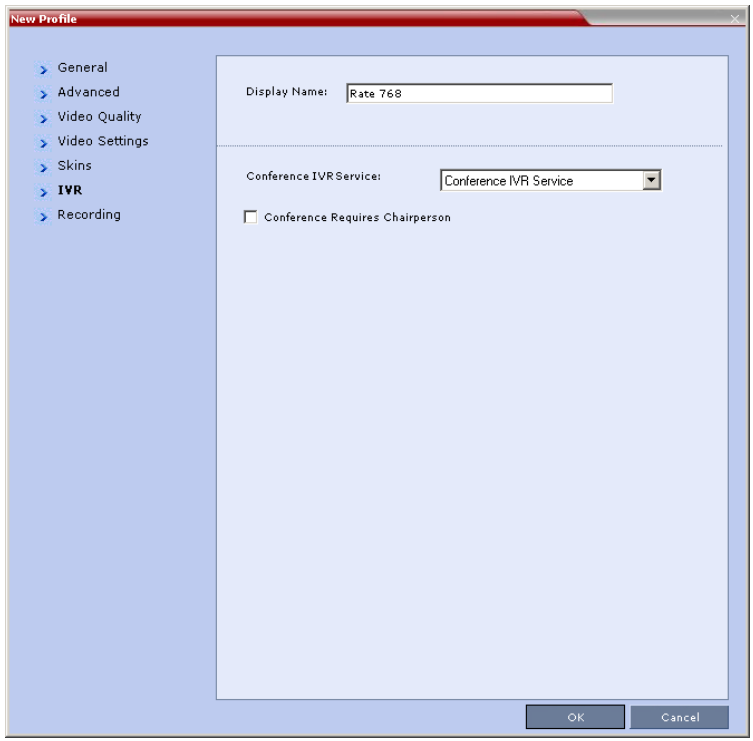


12 Select one of the *Skin* options.

When *Telepresence Mode* is enabled, the Skin options are disabled as the system uses a black background and the frames and speaker indication are disabled.

13 Click **IVR** tab.

The *New Profile - IVR* dialog box opens.



14 If required, set the following parameters:

Table 1-10 *Profile Properties - IVR*

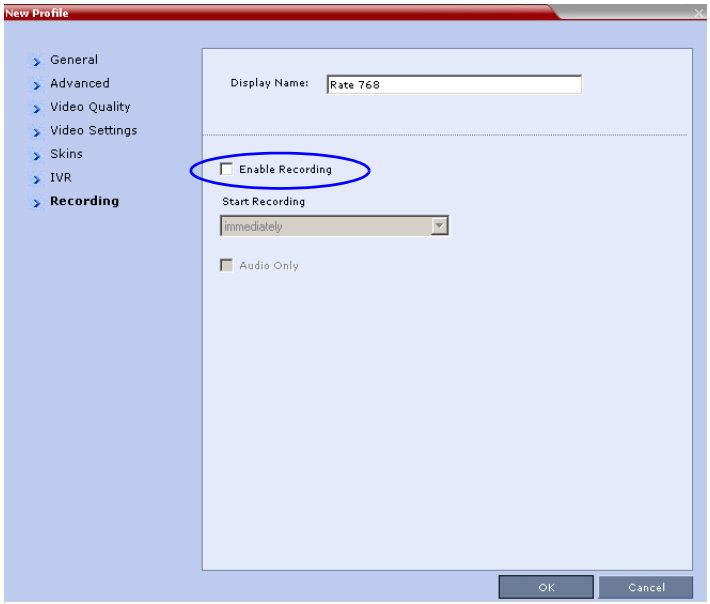
Field/Option	Description
Conference IVR Service	The default conference IVR Service is selected. You can select another conference IVR Service if required.

Table 1-10 *Profile Properties - IVR (Continued)*

Field/Option	Description
Conference Requires Chairperson	Select this option to allow the conference to start only when the chairperson connects to the conference and to automatically terminate the conference when the chairperson exits. Participants who connect to the conference before the chairperson are placed on *Hold* and hear background music (and see the *Welcome* video slide). Once the conference is activated, the participants are automatically connected to the conference. When the check box is cleared, the conference starts when the first participant connects to it and ends at the predefined time or according to the *Auto Terminate* rules when enabled.

15 **Optional**. Click the **Recording** tab to enable conference recording with *Polycom RSS 2000*.

16 Select the **Enable Recording** check box.



17 Define the following parameters:

Table 1-11 Profile Properties - Recording Parameters

Parameter	Description
Start Recording	Select one of the following: • **Immediately** – conference recording is automatically started upon connection of the first participant. • **Upon Request** – the operator or chairperson must initiate the recording (manual).
Audio Only	Select this option to record only the audio channel of the conference.

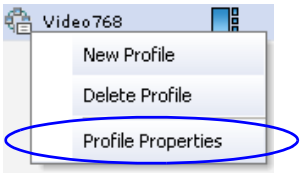
18 Click **OK** to complete the *Profile* definition.
A new *Profile* is created and added to the *Conference Profiles* list.

Modifying an Existing Profile

You can modify any of the Profile's parameters but you cannot rename the *Profile*.

To modify the Profile Properties:

1 In the *Conference Profiles* list, double -click the *Profile* icon or right-click the *Profile* icon, and then click **Profile Properties**.



The Profile *Properties - General* dialog box opens.

Deleting a Conference Profile

To delete a Conference Profile:

1. In the *Conference Profiles* list, select the *Conference Profile* you want to delete.
2. Click the **Delete Profile** ( ) button.
   or
   Right-click the *Conference Profile* to be deleted and select **Delete Profile** from the drop-down menu.

   A confirmation dialog box is displayed.
3. Click **OK** in the confirmation dialog box.
4. The *Conference Profile* is deleted.

A conference Profile cannot be deleted if it is being used by any conferencing entities such as Ongoing conferences, Meeting Rooms, Entry Queues, SIP Factories and Reservations.

2

Additional Conferencing Information

Various conferencing modes and video features require additional settings, such as system flag settings, conference parameters and other settings. In depth explanations of these additional settings are described in the following sections.

The RMX can function with two types of video Media Processing Modules (MPM): MPM and MPM+. These cards differ in their port capacity and their support of video resolutions. In addition to all video modes and features supported by MPM cards, MPM+ cards support additional video resolutions and video quality enhancement such as Video Clarity™ .

Video Session Modes

The RMX offers two video session modes: Continuous Presence and High Definition Video Switching. The video session type determines the video display options (full screen or split screen with all participants viewed simultaneously) and the method in which the video is processed by the MCU (with or without using the MCU's video resources).

Dynamic Continuous Presence (CP) Mode

The Continuous Presence mode offers 24 layouts to accommodate different numbers of participants and conference settings including support of the VUI annex to the H.264 protocol for endpoints that transmit wide video instead of 4CIF resolution.

For conferences with more participants than display squares, the RMX dynamic video mix capability allows the viewed sites to be modified throughout the conference. The displayed layout can be changed during an ongoing conference, allowing a participant to view different screen layouts of the other conference participants. These layout options allow conferences to have greater flexibility when displaying a large number of participants and maximizes the screen's effectiveness.

High Definition Video Switching Mode

In Video Switching mode all participants see the same video picture (full screen) and use only one CIF video resource for each connection. The current speaker is displayed in full screen on all the participants' endpoints, while the speaker sees the previous speaker. Switching between participants is voice-activated; whenever a participant starts to speak, he or she becomes the conference speaker and is viewed on all screens. All conference participants must use the same line rate and video parameters such as video protocol, frame rate, annexes and interlaced video mode as no video processing is performed.

High Definition Video Switching is an ultra-high quality video resolution enabling compliant endpoints to connect to conferences at resolutions of 1280x720 pixels (720p) and at line rates ranging from 384kbp to 4Mbps (with MPM) and 1920 x 1080 pixels (1080p) at line rates ranging from 384kbp to 6Mbps (with MPM+).

HD Video Switching uses less system resources. The maximum conference size is 80 video participants.

Continuous Presence (CP) Conferencing

Video quality in Continuous Presence mode is affected by the conference line rate (that determines the maximum line rate to be used by the connecting endpoints), and the video capabilities of the endpoints such as the video protocol, video resolution and frame rate.

The video protocol selected by the system determines the video compression standard used by the endpoints. In Continuous Presence conferences, the system selects the best video protocol for the endpoint according to its capabilities.

The following Video protocols are supported:

- **H.261** - the video compression algorithm mandatory to all endpoints. It is used by endpoints that do not support other protocols.
- **H.263** - a video compression algorithm that provides a better video quality than H.261. This standard is not supported by all endpoints.
- **H.264** - a video compression standard that offers improved video quality, especially at line rates lower than 384 Kbps.

Video Resolutions in CP

The RMX always attempts to connect to endpoints at the highest line rate. If the connection cannot be established, the RMX attempts to connect at the next highest line rate at its highest supported resolution.

The video resolution is also defined by the *Quality* settings:

- **Motion**, when selected, results lower video resolution.
- **Sharpness**, when selected, sends higher video resolution.

The combination of **frame rate** and **resolution** affects the number of video resources required on the MCU to support the call.

Table 2-1 *Video Resolution by Line Rate, Frame Rate & Video Resources*

Maximum Resolution	Minimum Line Rate Kbps	Maximum Frame Rate	Video Resources Required	
			MPM+	MPM
CIF	*128*	*30*	1	1
2CIF/WCIF	*256*	*30*	2.66	2
SD	*256*	*15*	2.66	2
*SD **	*256*	*30*	2.66	4
*SD **	*1024*	*60*	4	Not Supported
HD720p	*1024*	*30*	4	4
HD720p	*1920*	*60*	8	Not Supported
HD1080p	*4096*	*30*	8	Not Supported

* SD includes all resolutions above 2CIF (576 x 352 pixels) and below HD (720 x 1280 pixels)..

Additional Video Resolutions in MPM+ Mode

The following higher video quality resolutions are available when the RMX is working in *MPM+ Mode*:

- CIF 352 x 288 pixels at 50 fps.
- WCIF 512 x 288 pixels at 50 fps.
- WSD 848 x 480 pixels at 50 fps.
- W4CIF 1024 x 576 pixels at 30 fps.
- HD 720p 1280 x 720 pixels at 60 fps.
- HD 1080p 1920 x 1080 pixels at 30 fps.

The video resolution transmitted to any endpoint is determined by the endpoint's capabilities, the conference line rate, the *Conference Profile's Motion* and *Sharpness* settings and the RMX's *Card Configuration Mode (MPM* or *MPM+*).

Additional Intermediate Video Resolutions

Two higher quality, intermediate video resolutions replace the transmission of CIF (352 x 288 pixels) or SIF (352 x 240 pixels) resolutions to endpoints that have capabilities between:

- **CIF** (352 x 288 pixels) and **4CIF** (704 x 576 pixels) – the resolution transmitted to these endpoints is **432 x 336** pixels.
- **SIF** (352 x 240 pixels) and **4SIF** (704 x 480 pixels) – the resolution transmitted to these endpoints is **480 x 352** pixels.

The frame rates (depending on the endpoint's capability) for both intermediate resolutions are:

- In *MPM Mode* – 25 or 30 fps.
- In *MPM+ Mode* – 50 or 60 fps.

Video Display with CIF, SD and HD Video Connections

Although any combination of CIF, SD and HD connections is supported in all CP conferences, the following rules apply:

- In a 1X1 *Video Layout*:
  - **SD:** If the speaker transmits CIF, the MCU will send CIF to all participants, including the SD participants. In any other layout the MCU will transmit to each participant at the participant's sending resolution.
  - **HD:** The MCU transmits speaker resolution (including input from HD participants) at up to SD resolution. If 1x1 is the requested layout for the entire duration of the conference, set the conference to HD *Video Switching* mode.
- In asymmetrical *Video Layouts*:
  - **SD:** A participant in the large frame that sends CIF is displayed in CIF.
  - **HD:** Where participants' *video windows* are different sizes, the RMX transmits HD and receives SD or lower resolutions.
- In panoramic *Video Layouts*:
  - **SD**: Participants that send CIF also receive CIF.
  - **HD**: the RMX transmits HD and receives SD or lower resolutions, the RMX scales images from SD to HD resolution.

Setting the Maximum CP Resolution for Conferencing

The maximum CP resolution of the system is determined by the **MAX_CP_RESOLUTION** system flag. The default setting of the flag is **HD 1080.**

The flag values determine the maximum CP capability that each endpoint can declare:

Table 2-2 System Flag – MAX_CP_RESOLUTION Values

Flag Value	MPM	MPM+
	Each endpoint can declare capability up to:	
HD1080	HD	HD1080
HD720	HD	HD720

Table 2-2 System Flag – MAX_CP_RESOLUTION Values (Continued)

Flag Value	MPM	MPM+
	Each endpoint can declare capability up to:	
HD	HD	HD720
SD30	SD30	SD30
SD15	SD15	SD30
CIF	CIF	

For information about setting system flags, see "*System Configuration*" on page **16-10**.

CP Conferencing with H.263 4CIF

The video resolution of 4CIF in H.263 endpoints is only supported for conferences in which the video quality is set to sharpness and for line rates of 384 Kbps to 1920 Kbps as shown in Table 2-3.

Table 2-3 Video Quality vs. Line Rate

Endpoint Line Rate Kbps	Video Quality			
	Motion		Sharpness	
	Resolution	Frame Rate	Resolution	Frame Rate
128	QCIF	30	CIF	30
256	CIF	30	CIF	30
384 - 1920+	CIF	30	4CIF	15

The RMX Web Client supports monitoring of H.263 4CIF information. The H.245 or SDP tab includes the additional information.

The creation of a new H.263 4CIF slide is supported in the IVR Service in addition to the current H.263 IVR slide. If users utilize the default Polycom slides that are delivered with RMX 2000, the slide's resolution will be as defined in the profile, i.e. SD, HD, CIF, etc.... If users create a custom IVR slide, regardless of its resolution, the slide will be displayed in CIF resolution.

H.263 4CIF Guidelines

- H.263 4CIF is supported with H.323, SIP and ISDN connection endpoints.
- H.263 4CIF is supported in CP mode only. VSW is supported on the RMX 2000 in HD only.
- Click & View is supported in H.263 4CIF.
- AES encryption is supported with H.263 4CIF.
- H.263 4CIF is supported in recording by the RSS2000 and other recording devices.
- All video layouts are supported in H.263 4CIF, except 1x1 layout. In a 1x1 layout, the resolution will be CIF.
- For information about Resource Usage see Table 16-9 on page **16-37**.
- H.239 is supported in all 3 resolutions and based on the same bandwidth decision matrix as for HD.

High Definition Video Switching

High Definition Video Switching enables compliant endpoints to connect to conferences at resolutions of 1280x720 pixels (720p) at line rates ranging from 384kbp to 4Mb and 1920 x 1080 pixels (1080p) at line rates ranging from 384kbp to 6Mbps (with MPM+). Video display is in full screen mode only and video is switched to the speaker.

HD Video Switching uses less system resources than HD CP: only one CIF video resource per participant for HD resolution.

The supported conference size is up to 80 video participants and 120 voice participants.

High Definition Video Switching conferences require:

- All participants to have HD compliant endpoints.
- All participants to connect using the same conference line rate.

High Definition Video Switching conferencing mode is unavailable to ISDN participants.

The recommended number of connections at *HD1080p* resolution in an RMX with two *MPM+* cards is:

- 160 participants at line rates of up to 2Mbps
- 80 participants at line rates of up to 4Mbps
- 40 participants at line rates of up to 6Mbps

HD VSW Guidelines

- The display aspect ratio is 4x3 or 16x9.
- Site names, skins, etc... are not supported in HD Video Switching.
- Video forcing is enabled at the conference and participant levels.
- Endpoints that do not support HD or are unable to meet these requirements connect as Secondary (audio only).

Enabling HD Video Switching

For the MCU to run HD Video Switching conferences the following criteria must be met:

- The *HD_THRESHOLD_BITRATE* flag must be set in the *System Configuration*. The value of this flag is the **system** minimum threshold bit rate.

The HD_THRESHOLD_BIT RATE flag is responsible for negotiation only, It does not guarantee that the endpoint will open an HD channel or transmit on an opened HD channel.

- The line rate selected in the conference Profile must be the same as or higher than that specified by the *HD_THRESHOLD_BITRATE* flag.
- The *High Definition Video Switching* option must be selected in the profile. For more information see "*Defining Profiles*" on page **1-8**.
- The RMX 2000 must have available resources (ports).
- The endpoints must support HD.

Modifying the HD Video Switching Threshold Bit Rate

To Modify the HD Video Switching Threshold:

1. Click **Setup>System Configuration**.
   The *System Flags* dialog box opens.
2. Set the **HD_THRESHOLD_BITRATE** flag to the required line rate value (range 384kbps to 4Mbs, default is 768 Kbps).
3. Click **OK**.

   The MCU must be reset the MCU for flag changes to take effect.

   For more information see "*System Configuration*" on page **16-10**.

Creating a High Definition Video Switching Profile

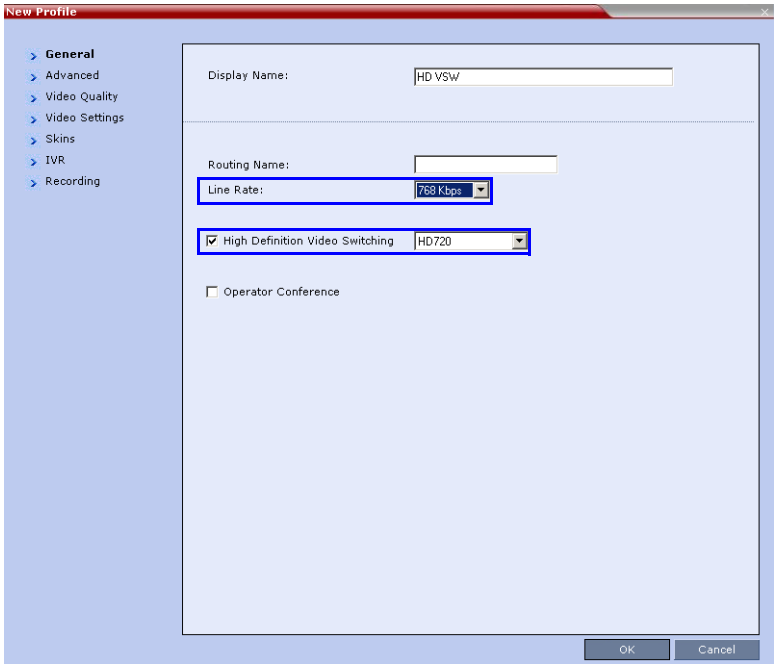
An HD Video Switching enabled Profile must be created prior to running HD Video Switching conferences.

High Definition Video Switching conferences, Entry Queues and Meeting Rooms are created by selecting an HD Video Switching-enabled Profile and must be set to the same line rate as the target conference.

To connect to an HD Video Switching conference via an Entry Queue, the Entry Queue must be HD Video Switching enabled. It is recommended to use the same Profile for both the target conference and Entry Queue.

To Create an HD Enabled Profile:

1. In the *New Profile – General* tab, in the *Line Rate* field, enter a bit rate that is higher than the defined HD threshold.
2. Select the **High Definition Video Switching** check box.

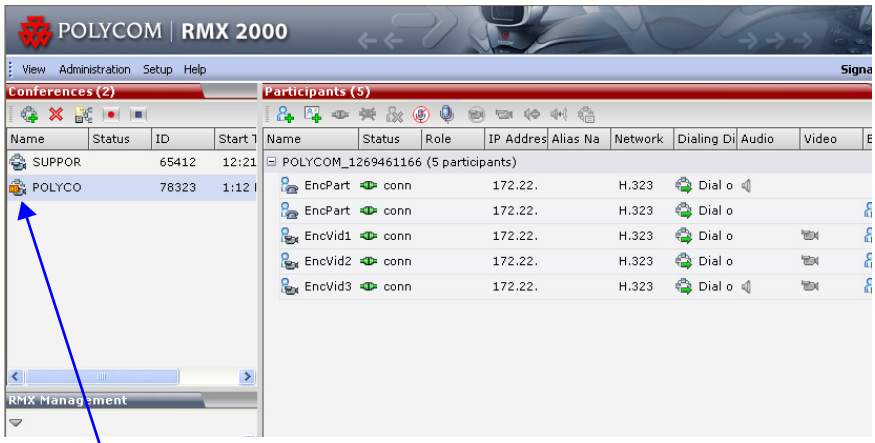


3. Select the resolution for the conference: **HD720** or **HD1080** (with MPM+).
4. Click **OK**.

   For more information, see "*Defining Profiles*" on page **1-8**.

Monitoring High Definition Video Switching Conferences

HD conferences appear with the HD ( ) icon in the conferences list to indicate the currently running HD conference(s).


POLYCOM | RMX 2000
View Administration Setup Help
Conferences (2)
Participants (5)
Name Status ID Start T
SUPPOR 65412 12:21
POLYCO 78323 1:12
Name Status Role IP Addres Alias Na Network Dialing Di Audio Video
POLYCOM_1269461166 (5 participants)
EncPart conn 172.22. H.323 Dial o
EncPart conn 172.22. H.323 Dial o
EncVid1 conn 172.22. H.323 Dial o
EncVid2 conn 172.22. H.323 Dial o
EncVid3 conn 172.22. H.323 Dial o
RMX Management
HD Conference

Monitoring is done in the same way as for standard conferences.

For more information, see "*Conference and Participant Monitoring*" on page **9-1.**

H.239

The H.239 protocol allows compliant endpoints to transmit and receive two simultaneous video streams:

- **People Conference –** Continuous Presence or Video Switched conference
- **Content Conference –** Video Switching conference for content sharing

By default, all conferences, Entry Queues, and Meeting Rooms launched on the RMX 2000 have H.239 capabilities.

To view Content, endpoints must use the same Bit Rate, Protocol, and Resolution. An endpoint may not send Content while connecting to an Entry Queue.

Endpoints without H.239 capability can connect to the video conference without Content.

Cascade links declare H.239 capabilities and they are supported in Star and MIH cascading topologies. For more details, see "*Cascading Conferences - H.239-enabled MIH Topology*" on page **2-59**.

Content Transmission Modes

The Content channel can transmit one of the following modes:

- **Graphics –** default mode, for standard graphics
- **Hi-res Graphics –** requiring a higher bit rate, for high quality display or highly detailed graphics
- **Live Video –** highest bit rate, for video clips or live video display

The highest common Content bit rate is calculated for the conference each time an endpoint connects. Therefore, if an endpoint connects to an ongoing conference at a lower bit rate than the current bit rate, the Content bit rate for the current conference is re-calculated and decreased.

Bit rate allocation by the MCU is dynamic during the conference and when the Content channel closes, the video bit rate of the *People conference* is restored to its maximum.

During a conference the MCU will not permit an endpoint to increase its bit rate, it can however change its Content resolution. The RMX can decrease the allocated Content bit rate during a conference.

The following table shows the bit rate allocated to the Content channel from the video channel in each of the three Content modes:

Table 2-4 Bit Rate Allocation to Content Channel

Conf Kbps / Mode	64/ 96	128	256	384	512	768	1024	1472/1920/ 4096-VSW
Graphics	0	64	64	128	128	256	256	256
Hi-res Graphics	0	64	128	192	256	384	384	512
Live Video	0	64	128	256	384	512	768	768

Content Protocol

H.263 Annex T and H.264 protocols are supported for the Content transmission.

H.264 provides higher video quality at video resolutions of up to HD.

Endpoint Capabilities

- If an endpoint that supports only *H.263* for Content Sharing connects to a conference with an *Up to H.264* Content sharing Profile:
  - *H.263* is used for Content if that participant is the first to connect to the conference
  - Content sharing is stopped for all participants if the connection occurs after Content sharing has started. When Content sharing is restarted by the user, Content is shared using *H.263*.
- If an endpoint that does not support *H.264* Content sharing disconnects from a conference with an *Up to H.264* Content Sharing Profile, the Content sharing continues using *H.263.* This is true even if all the remaining connected endpoints support *H.264*. If Content sharing is stopped and restarted by the user, Content sharing is automatically upgraded to use *H.264*.
- The *H239_FORCE_CAPABILITIES System Flag* in *system.cfg* gives additional control over Content sharing.

  When the flag is set to *NO*, the RMX only verifies that the endpoint supports the content protocols: *Up to H.264* or *H.263*.

When set to *YES*, the RMX checks frame rate, bit rate, resolution, annexes and all other parameters of the Content mode as declared by an endpoint during the capabilities negotiation phase. If the endpoint does not support the Content capabilities of the MCU the participant will not be able to send or receive content over a dedicated content channel. The flag's default value is NO.

If the *System Flag*, does not exist in the system, it must be created using the *RMX Menu – Setup* option. For more information see the *RMX 2000 Administrator's Guide*, "*Modifying System Flags*" on page **16-10**.

Entry Queues

- The selection of either *H.263* or *Up to H.264* in the Entry Queue Profile does not affect how Content is shared.
- When the endpoint is moved to the conference from the Entry Queue, the endpoint shares Content according to the guidelines set out under *Endpoint Capabilities* and according to the content protocol that is defined for the target conference.

Cascade Links

Content is shared across a Cascaded Link using *H.263* irrespective of whether either or both the cascade-enabled Entry Queue and the Cascaded Link have *Up to H.264* Content sharing defined in their profiles.

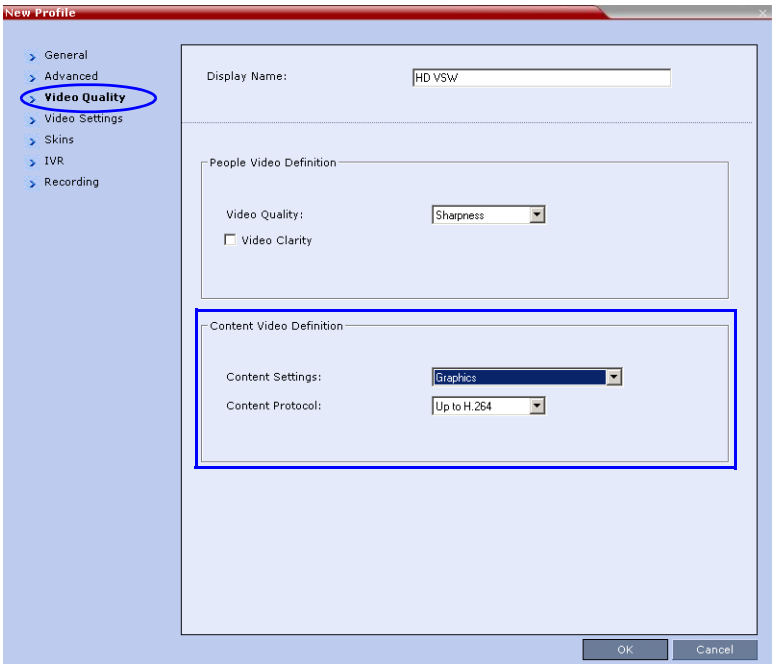
Defining Content Sharing Parameters for a Conference

To define Content Sharing Parameters:

1. In the *RMX Management* pane, click **Conference Profiles.**
2. In the *Conference Profiles* pane, click **New Profile**.

   The *New Profile – General* dialog box opens.
3. Define the Profile General parameters.
4. Optional. Click the **Advanced** tab and define additional conference parameters.

5 Click the **Video Quality** tab.



6 In the *Content Video Definition* section, select the *Content Settings* and *Protocol* as follows:

Table 2-5 *H.239 Content Options*

Field	Description
Content Settings	Select the transmission mode for the Content channel: • ***Graphics*** — basic mode, intended for normal graphics • ***Hi-res Graphics*** — a higher bit rate intended for high resolution graphic display • ***Live Video*** — Content channel displays live video Selection of a higher bit rate for the Content results in a lower bit rate for the people channel.

Table 2-5 H.239 Content Options (Continued)

Field	Description
Content Protocol	**H.263** – Content is shared using *H.263* even if some endpoints have *H.264* capability. **Up to H.264** – *H.264* is the default Content sharing algorithm. When selected: • Content is shared using *H.264* if all endpoints have *H.264* capability. • Content is shared using *H.263* if all endpoints do not have *H.264* capability. • Endpoints that do not have at least *H.263* capability can connect to the conference but cannot share Content.

7 Click **OK.**

Sending Content to Legacy Endpoints

The RMX can be configured to send Content to H.323/SIP/ISDN endpoints that do not support H.239 Content (legacy endpoints) over the video (people) channel, allowing the participants using the legacy endpoints to view the Content as the other conference participants.

Guidelines for Sending Content to Legacy Endpoints

- This option is enabled only in **MPM+** *Card Configuration Mode* and *Resource Allocation Mode* is set to **Flexible Mode**.
- This option is valid when sending Content as a separate stream is enabled in the *System Configuration* and the flag: ENABLE_H239 is set to YES.
- One additional SD (3CIF) video resource is allocated to the conference when Content is sent to legacy endpoints. The allocation is done only when a legacy endpoint is connected to the conference and a Content session is initiated and transmitted via the video channel.
  Once the resource is allocated, it remains allocated to the conference until the conference ends.

If the system cannot allocate the resource required for sending the Content, the conference status changes to "Content Resource Deficiency" and Content will not be sent to the legacy endpoints.

As the resource required for sending Content to legacy endpoints is allocated on the fly, when scheduling a reservation, in rare occasions when the MCU is fully loaded, "Resource deficiency" may be encountered. This may prevent participants from connecting to the conference or from Content being sent to the legacy endpoint. To ensure resource for sending Content to legacy endpoints, add one resource to the number of resources defined in the *Reserve Resources for Video Participants* field, in the *Conference Properties - General* dialog box.

- Non-H.239 (legacy) endpoints receive the Content via the video channel using the same video protocol and resolution with which they receive video.
- The highest Content resolution is HD720p, even if HD1080p is selected in the Profile.
- Content cannot be sent to legacy endpoints when *Same Layout* mode is selected for the conference.
- This option in not supported in *High Definition Video Switching* conferences.
- When Content is transmitted, the Site Name of the endpoints cannot be viewed.
- Content can be sent to legacy endpoints in gateway calls.
- When moving a legacy participant to the *Operator conference*, Content will not be available to the legacy endpoint.

Interoperability with Polycom CMA and DMA

The CMA uses the Profiles that are stored in the RMX. If the *Send Content to Legacy Endpoints* option is enabled in the Conference Profile, this option will be enabled in the conference started from the CMA that uses that Profile. However, the CMA does not display an indication that this option is enabled for the conference.

A new conference can be started on the DMA using a Conference Profile that is defined on the RMX or by defining all the conference parameters. The *Send Content to Legacy Endpoints* option can be enabled only in the Conference Profile defined in the RMX, therefore, to include this option in the conference started on the DMA use an RMX existing Profile. However, the DMA does not display an indication that this option is enabled for the conference.

Content Display on Legacy Endpoints

When Contents is sent to legacy endpoints, their video layout automatically changes to a "Content layout" which is defined by the system flag LEGACY_EP_CONTENT_DEFAULT_LAYOUT and the Content is shown in the larger/top left ("speaker") window. The video layouts of the other conference participants do not change.

The switch to the Content layout occurs in the *Auto Layout*, *Presentation Mode*, *Lecture Mode* and when a layout is selected for the conference. However, in *Lecture Mode*, when Content is sent to legacy endpoints, when switching to the Content layout, the Content is shown in the "lecturer/speaker" window and the lecturer is show in a second window. If the layout contains more than two windows, all other windows will be empty. All other participants will see the lecturer in full screen.

In *Same Layout* mode, Content cannot be sent to legacy endpoints.

The LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag default is set to a layout of 1+4 where the Content is shown in the large window and the other conference participants are shown in the small windows. This default value can be changed in the *System Configuration*.

When Content is stopped, the layout of the legacy participants returns to the last video layout seen prior to the Content mode.

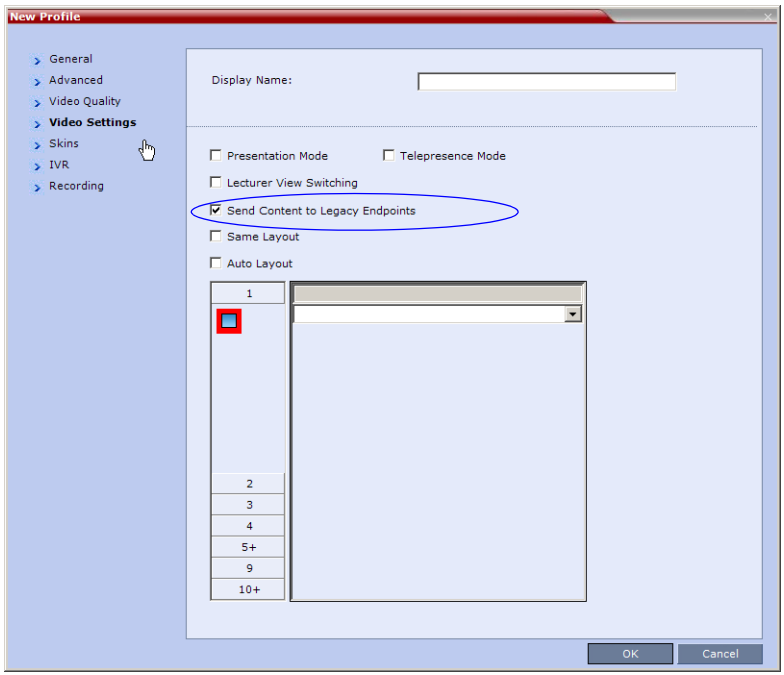
The Legacy participants can change their layout using *Click&View*. In such a case, the Content is forced to the "speaker" window.

The RMX user can also change the layout for the participants the legacy endpoints (selecting personal layout).

When forcing a video participant to the Content window (instead of Content), the Content display can be restored only by selecting any other video layout.

Enabling the Send Content to Legacy Endpoints Option

The **Send Content to Legacy Endpoint** option is enabled in the *Conference Profile - Video Settings* tab. It is selected by default.



If *High Definition Video Switching* option is selected in the *Conference Profile - General* tab, the *Send Content to Legacy Endpoint*s selection is cleared and the option is disabled.

If the *Same Layout* option is selected in the *Conference Profile - Video Settings* tab, the *Send Content to Legacy Endpoints* selection is cleared and is disabled.

Changing the Default Layout for Displaying Content on Legacy Endpoints

The default layout that will be used to display Content on the screens of legacy endpoints is defined by the system flag **LEGACY_EP_CONTENT_DEFAULT_LAYOUT.**

The configured default layout is 1+4 (CP_LAYOUT_1P4VER). You can change the default layout configuration by entering a new value for the flag in the system configuration.

To modify system flags:

1. On the *RMX 2000* menu, click **Setup > System Configuration**.
   The *System Flags* dialog box opens.
2. In the *MCMS_PARAMETERS* tab, click the **LEGACY_EP_CONTENT_DEFAULT_LAYOUT** entry.
   The *Update Flag* dialog box is displayed.
3. In the *Value* field, enter the flag value for the required layout as follows:

Table 2-6 *LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values*

Layout	Flag Value
	CP_LAYOUT_1X1
	CP_LAYOUT_1X2
	CP_LAYOUT_1X2HOR
	CP_LAYOUT_1X2VER
	CP_LAYOUT_2X1
	CP_LAYOUT_1P2HOR

Table 2-6 *LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values (Continued)*

Layout	Flag Value
	CP_LAYOUT_1P2HOR_UP
	CP_LAYOUT_1P2VER
	CP_LAYOUT_2X2
	CP_LAYOUT_1P3HOR_UP
	CP_LAYOUT_1P3VER
	CP_LAYOUT_1P4HOR_UP
	CP_LAYOUT_1P4HOR
	CP_LAYOUT_1P4VER
	CP_LAYOUT_1P5
	CP_LAYOUT_1P7
	CP_LAYOUT_1P8UP
	CP_LAYOUT_1P8CENT
	CP_LAYOUT_1P8HOR_UP
	CP_LAYOUT_3X3

Table 2-6 LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values (Continued)

Layout	Flag Value
	CP_LAYOUT_2P8
	CP_LAYOUT_1P12
	CP_LAYOUT_4X4

4 Click **OK**.
The flag is updated in the *MCMS_PARAMETERS* list.

5 Click **OK**.

For flag changes to take effect, reset the MCU. For more information, see the *RMX Administrator's Guide*, "*Resetting the RMX"* on page **16-90**.

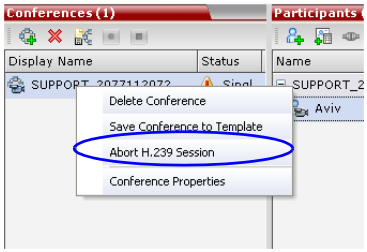
Stopping a Content Session

In some cases, when one participant ends the Content session from his/her endpoint, the Content token is not released and other participants cannot send Content.

The RMX User can withdraw the Content token from the current holder and to return it to the MCU for assignment to other endpoints.

To end the current Content session:

- In the *Conferences* list pane, right-click the conference icon and then click **Abort H.239 Session**.


Conferences (1)
Participants
Display Name
Status
Name
Delete Conference
Save Conference to Template
Abort H.239 Session
Conference Properties
Aviv

Lecture Mode

Lecture Mode enables all participants to view the lecturer in full screen while the conference lecturer sees all the other conference participants in the selected layout while he/she is speaking. When the number of sites/endpoints exceeds the number of video windows in the layout, switching between participants occurs every 15 seconds.

Automatic switching is suspended when one of the participants begins talking, and it is resumed automatically when the lecturer resumes talking.

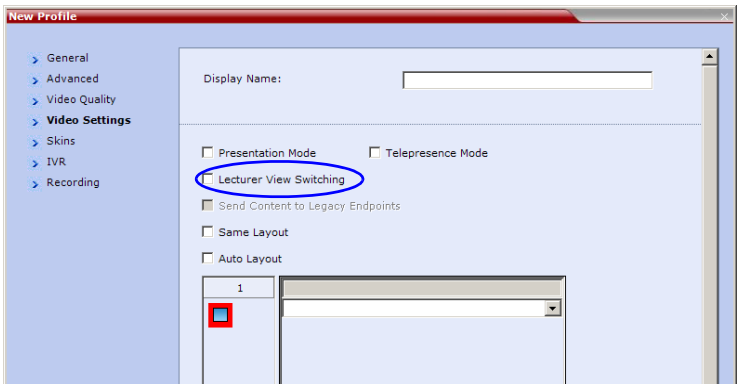
Enabling Lecture Mode

Lecture Mode is enabled at the conference level by selecting the lecturer.

Automatic switching between participants viewed on the lecturer's screen is enabled in the conference Profile.

Enabling the Automatic Switching

- In the *Profile Properties - Video Settings* dialog box, select the **Lecturer View Switching** check box.



This option is activated when the conference includes more sites than windows in the selected layout. If this option is disabled, the participants will be displayed in the selected video layout without switching.

For more information about Profile definition, see "*Defining Profiles*" on page **1-8**.

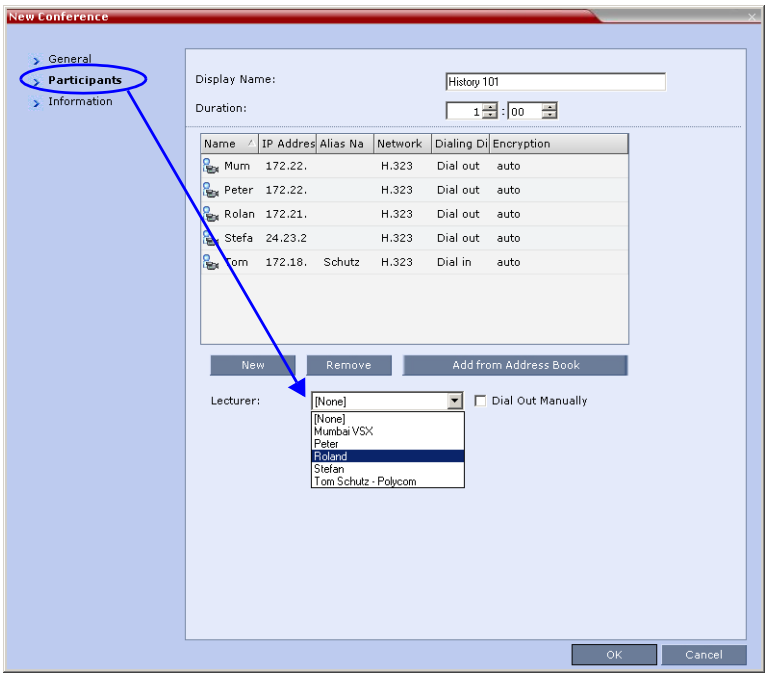
Selecting the Conference Lecturer

A conference can be set to Lecture Mode when:

- Defining a new ongoing Conference, after the adding or defining the participant to be designated as lecturer
- During an ongoing conference, after the participant (to be designated as lecturer) has connected to the conference.

To enable Lecture Mode for the Conference:

- In the *Conference Properties - Participant* dialog box, select the **Lecturer** from the list.



Lecture Mode Monitoring

A conference in which the Lecture Mode is enabled is started as any other conference. The conference runs as an audio activated Continuous Presence conference until the lecturer connects to the conference. The selected video layout is the one that is activated when the conference starts. Once the lecturer is connected, the conference switches to the Lecture Mode.

When *Lecturer View Switching* is activated, it enables automatic switching between the conference participants in the lecturer's video window. The switching in this mode is not determined by voice activation and is initiated when the number of participants exceeds the number of windows in the selected video layout. In this case, when the switching is performed, the system refreshes the display and replaces the last active speaker with the current speaker.

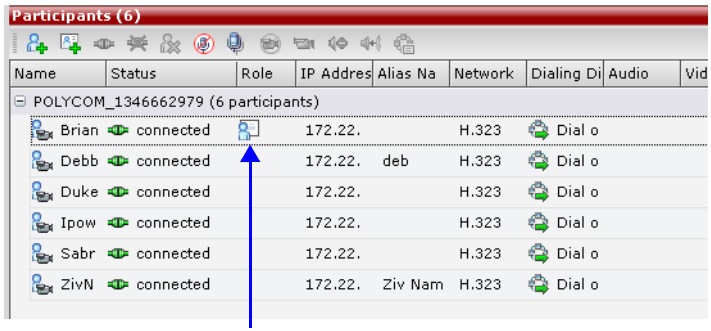
When one of the participants is talking, the automatic switching is suspended, showing the current speaker, and it is resumed when the lecturer resumes talking.

If the lecturer is disconnected during an Ongoing Conference, the conference resumes standard conferencing.

Forcing is enabled at the Conference level only. It applies only to the video layout viewed by the lecturer as all the other conference participants see only the lecturer in full screen.

If an asymmetrical video layout is selected for the lecturer (i.e. 3+1, 4+1, 8+1), each video window contains a different participant (i.e. one cannot be forced to a large frame and to a small frame simultaneously).

When *Lecture Mode* is enabled for the conference, the lecturer is indicated by an icon ( ) in the *Role* column of the *Participants* list.

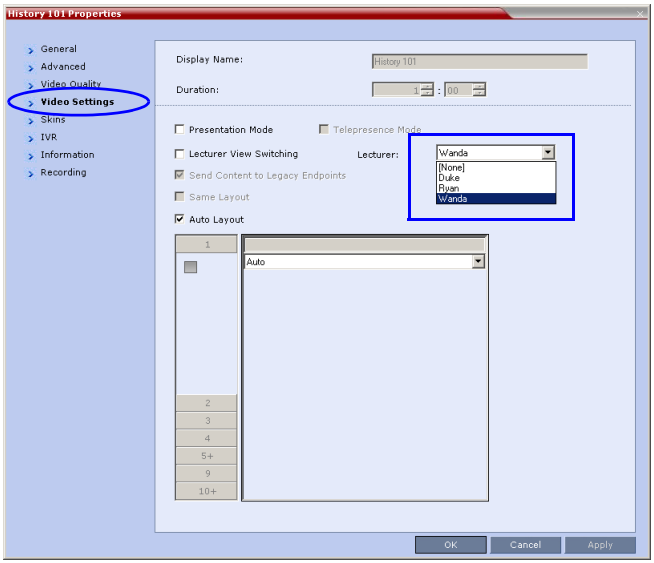


Participant designated as the Lecturer

To control the Lecture Mode during an Ongoing Conference:

During the Ongoing Conference, in the *Conference Properties - Video Settings* dialog box you can:

- Enable or disable the Lecture Mode and designate the conference lecturer in the *Lecturer* list; select **None** to disable the Lecture Mode or select a participant to become the lecturer to enable it.
- Designate a new lecturer.



- Enable or disable the Lecturer View Switching between participants displayed on the lecturer monitor window by selecting or clearing the *Lecturer View Switching* check box.
- Change the video layout for the lecturer by selecting another video layout.

Closed Captions

Endpoints can provide real-time text transcriptions or language translations of the video conference by displaying closed captions. The captions for a conference may be provided by the captioner who is present in the conference, or the captioner may use a telephone or web browser to listen to the conference audio. When the captioner sends a unit of text, all conference participants see it on the main monitor for 15 seconds. The text then disappears automatically.

The captioner may enter caption text using one of the following methods:

- Remotely, via a dial-up connection to the system's serial RS-232 port.
- In the room using equipment connected directly to the serial port.
- In the room or remotely, using the Polycom HDX web interface.

Closed Captions Guidelines

- The Captions display properties are configured on the endpoint sending the captions.
- Closed Captions content is defined from the endpoint. The RMX only transmits it to the endpoints.
- When enabled, Closed Captions are available to all endpoints supporting FECC.
- Closed Captions are supported in H.323 and SIP connections.
- The FECC indications during ongoing conferences are used when closed captions are active.
- When Closed Captions are enabled, muting an endpoint may cause the display of the "Far Mute" indication on all the screens of the endpoints connected to the conference.
- The Closed Captions option is not supported in cascading conferences (they can only be viewed in the local conference) as FECC is not supported in cascading links.
- Site name display is not affected by closed captions display.
- Closed Captions are supported by the RMX in the following configurations and conferencing modes:
  - MPM and MPM+ *Card Configuration Modes*.
  - *High Definition Video Switching* and Continuous Presence conferencing modes.

— Encrypted and non-encrypted conferences.
— Conferences with Content.

Enabling Closed Captions

Closed Captions are enabled by a system flag. By default, Closed Captions are disabled.

To change the flag value:

1. On the *RMX 2000* menu, click **Setup > System Configuration**.
   The *System Flags* dialog box opens.
2. In the *MCMS_PARAMETERS* tab, click the **New Flag** (

) button.
   The *New Flag* dialog box is displayed.
3. In the *New Flag* field enter **ENABLE_CLOSED_CAPTION**.
4. In the *Value* field enter **YES** to enable *Closed Captions* or **NO** to disable their display.
5. Click **OK** to close the *New Flag* dialog box.
   The new flag is added to the flags list.
6. Click **OK** to close the *System Flags* dialog box.

For flag changes to take effect, reset the MCU. For more information, see the *RMX Administrator's Guide*, "*Resetting the RMX*" on page **16-90**.

Media Encryption

Encryption is available at the conference and participant levels, based on AES 128 Media Encryption and DH 1024 Key Exchange standards.

Media Encryption guidelines:

- Encryption is not available in all countries and it is enabled in the MCU license. Contact Polycom Support to enable it.
- Endpoints must support both AES 128 encryption and DH 1024 key exchange standards which are compliant with H.235 (H.323) to encrypt and to join an encrypted conference.
- The encryption mode of the endpoints is not automatically recognized, therefore the encryption mode must be set for the conference or the participants (when defined).
- Conference level encryption must be set in the Profile, and cannot be changed once the conference is running.
- Mixing encrypted and non-encrypted endpoints in one conference is supported only for H.323 **defined** participants. Different states of encryption for predefined IP participants are possible, based on system flag settings: (ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF).
- In Cascaded conferences, to encrypt the conference the link between the cascaded conferences must be encrypted.
- The ISDN/PSTN (H.320) protocol, as well as ISDN/PSTN endpoints do not support encryption. ISDN/PSTN participants can therefore only connect to encrypted conferences if the system is set up to allow the mixing of encrypted/non-encrypted participants in the same conference.

Conference Access

You can define whether access to conferences for encrypted and non-encrypted IP participants is done at the conference level or at the participant level.

Conference access is set in the system configuration by the flag ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF as follows:

Table 2-7 Participant Level Encryption According to Flag Settings

Flag Settings	Encryption is enabled in Profile	Encryption is disabled in the Profile
YES	Encryption is determined at the participant level, therefore encrypted and non-encrypted participants can join the conference.	Encryption is determined at the participant level, therefore encrypted and non-encrypted participants can join the conference.
NO	Encryption is determined at the conference level, therefore only encrypted participants can join the conference.	Encryption is determined at the conference level, therefore only non-encrypted participants can join the conference. Encrypted participants cannot be moved from an encrypted Entry Queue to this conference.

Encrypted Entry Queue Access

To be able to join a conference from an Entry Queue as an encrypted participant, encryption must be enabled in the Profile assigned to the Entry Queue. All non-encrypted participants connecting to an encrypted Entry Queue are disconnected from the MCU.

Encrypted participants can be moved from an encrypted Entry Queue to the destination conference depending on the destination conference settings and the setting of the flag in the system configuration.

Table 2-8 summarizes the dependencies between the connecting participant's defined encryption status, the Conference/Entry Queue encryption setting and the flag setting.

Table 2-8 *Participant Level Encryption Options*

Participant Encryption Field	Conference/Entry Queue Encryption Enabled		Conference/Entry Queue Encryption Disabled	
	Participant Level Flag		Participant Level Flag	
	YES	NO	YES	NO
Auto	Joins conference encrypted	Joins conference encrypted	Joins conference non-encrypted	Joins conference non-encrypted
On	Joins conference encrypted	Joins conference encrypted	Joins conference encrypted	Cannot join conference
Off	Joins conference non-encrypted **Note:** Enables ISDN participants to join an encrypted conference	Cannot join conference	Joins conference non-encrypted	Joins conference non-encrypted

Move Guidelines

- When participants are moved to another conference their encryption settings are evaluated to determine if the move is permitted. If not, the move fails and the participants remain in their original conference.
- When the flag is set to YES, participants can move between conferences that have different encryption settings. For example, encrypted participants can move to encrypted and non-encrypted conferences.
- When the flag is set to NO, the participant's encryption setting must match the conference encryption setting to be moved to the other conference. For example, encrypted participants can move only from an encrypted conference to another encrypted conference.

Encryption Flag Settings

To modify the Encryption flag:

1. Click **Setup>System Configuration**.
   The *System Flags* dialog box opens.
2. Set the **ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF** flag to **YES** or **NO**.
3. Click **OK**.

For more information, see "*System Configuration*" on page **16-10**.

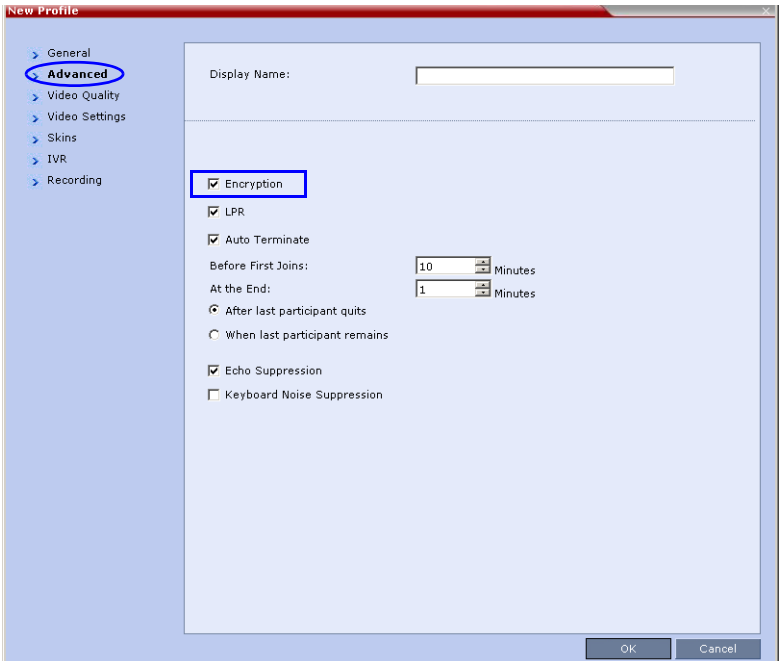
- Reset the MCU for flag changes to take effect.

Enabling Encryption in the Profile

Encryption for the conference is in the Profile and cannot be changed once the conference is running.

To enable encryption at the conference level:

- In the *Conference Profile Properties – Advanced* dialog box, select the **Encryption** check box.

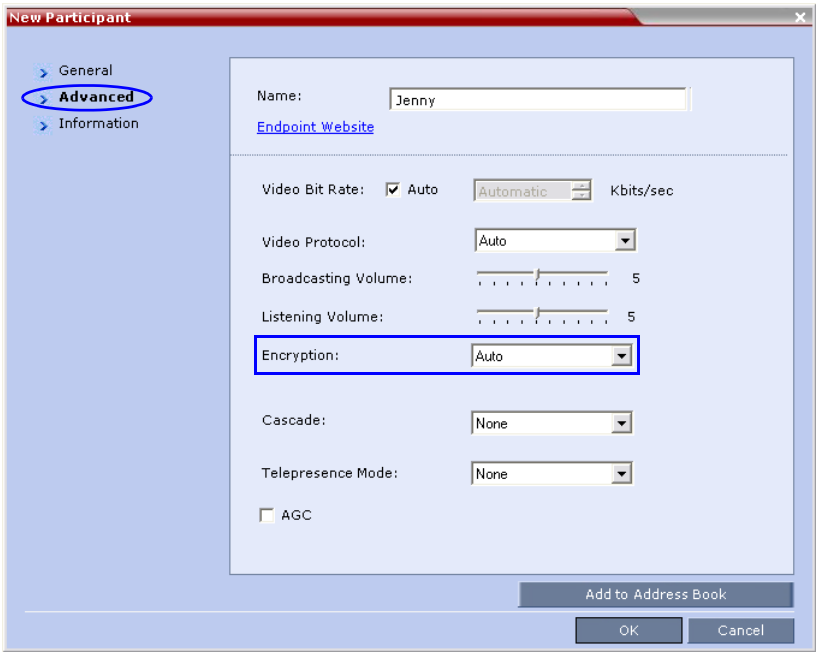


Enabling Encryption at the Participant Level (IP Only)

You can select the encryption mode for each of the defined participants. Encryption options are affected by the settings of the flag in the system configuration. Undefined participants are connected with the Participant *Encryption* option set to **Auto**, inheriting the conference/Entry Queue encryption setting.

To enable encryption at the participant level:

- In the *Participant Properties – Advanced* dialog box, in the *Encryption* list, select one of the following options: **Auto**, **On**, or **Off**.



— **Auto** - The participant inherits the conference/Entry Queue encryption setting. The participant connects as encrypted only if the conference is defined as encrypted.

— **Yes** - The participant joins the conference/Entry Queue is *encrypted*.

— **No** - The participant joins the conference/Entry Queue is *non-encrypted*.

Monitoring the Encryption Status

The conference encryption status is indicated in the *Conference Properties - General* dialog box.

The participant encryption status is indicated by a check mark in the *Encryption* column in the *Participants* list pane.

An encrypted participant who is unable to join a conference is disconnected from the conference. The disconnection cause is displayed in the *Participant Properties – Connection Status* tab, *Security Failure* indication, and the *Cause* box identifies the encryption related situation.

For more information about monitoring, see "*Conference and Participant Monitoring*" on page **9-1**.

LPR – Lost Packet Recovery

Lost Packet Recovery (*LPR*) and *Dynamic Bandwidth Allocation* (*DBA*) help minimize media quality degradation that can result from packet loss in the network.

Packet Loss

Packet Loss refers to the failure of data packets, transmitted over an IP network, to arrive at their destination. *Packet Loss* is described as a percentage of the total packets transmitted.

Causes of Packet Loss

Network congestion within a LAN or WAN, faulty or incorrectly configured network equipment or faulty cabling are among the many causes of Packet Loss.

Effects of Packet Loss on Conferences

Packet Loss affects the quality of:

- **Video –** frozen images, decreased frame rate, flickering, tiling, distortion, smearing, loss of lip sync
- **Audio –** drop-outs, chirping, audio distortion
- **Content –** frozen images, blurring, distortion, slow screen refresh rate

Lost Packet Recovery

The *Lost Packet Recovery* (*LPR*) algorithm uses *Forward Error Correction* (*FEC*) to create additional packets that contain recovery information. These additional packets are used to reconstruct packets that are lost, for whatever reason, during transmission. *Dynamic Bandwidth Allocation* (*DBA*) is used to allocate the bandwidth needed to transmit the additional packets.

Lost Packet Recovery Guidelines

- *LPR* is supported in H.323 networking environments only.
- In *LPR*-enabled *Continuous Presence* conferences:
  - Both *LPR*-enabled and non *LPR*-enabled endpoints are supported.
  - The *LPR* process is not applied to packet transmissions from non *LPR*-enabled H.323, SIP and H.320 endpoints.
- In *LPR*-enabled *Video Switched* conferences:
  - SIP and H.320 endpoints are not supported.
  - Cascaded links to MGC are not supported.
  - Non H.323 participants cannot be created, added or moved to *LPR*-enabled *Video Switched* conferences.
- When connecting via an *Entry Queue:*
  - A participant using an *LPR*-enabled endpoint cannot be moved to a non *LPR*-enabled conference.
  - SIP and H.320 participants cannot be moved to *LPR*-enabled *Video Switched* conferences.
- If packet loss is detected in the packet transmissions of either the video or Content streams:
  - *LPR* is applied to both the video and Content streams.
  - *DBA* allocates bandwidth from the video stream for the insertion of additional packets containing recovery information.

Enabling Lost Packet Recovery

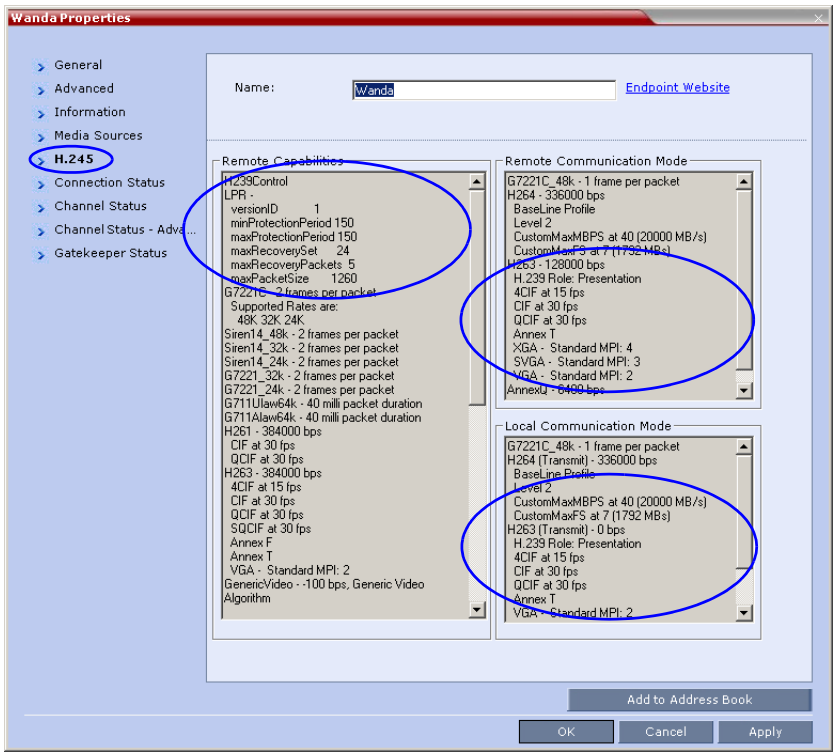
LPR is enabled or disabled in the *Conference Profile* dialog box.

- **CP Conferences –** *LPR* is enabled by default in the *New Profile – Advanced* dialog box.
- **HD VSW Conferences –** If *High Definition Video Switching* is selected, the LPR check box is automatically cleared and *LPR* is disabled. *LPR* can be enabled for HD VSW conferences but H.320 and SIP participants will not be able to connect.

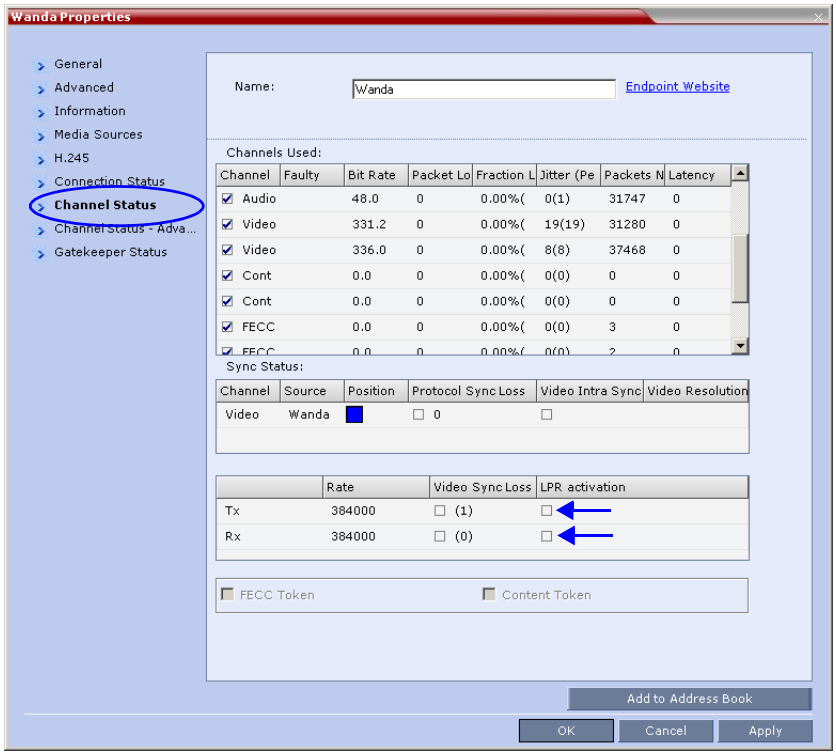
  For more information, see "*Defining Profiles*" on page **1-8**.

Monitoring Lost Packet Recovery

In the *Participant Properties – H.245* tab, *LPR* activity is displayed in all three panes.



In the *Participant Properties – Channel Status* tab, check box indicators show *LPR* activation in the local and remote (transmit and receive) channels.



Telepresence Mode

RMX 2000 supports the Telepresence Mode allowing multiple participants to join a telepresence conference from RPX and TPX high definition rooms as well as traditional, standard definition video conferencing systems.

TPX (Telepresence) and RPX (Realpresence) room systems are configured with high definition cameras and displays that are set up to ensure that all participants share a sense of being in the same room.

Figure 2-1 *Realpresence Participants using two RPX HD 400 Room Systems*

The following are examples of situations where an RMX is needed for *Telepresence* configurations:

- RPX to TPX
- RPX 2-cameras/screens to RPX 4-cameras/screens
- 3 or more RPXs
- 3 or more TPXs

RMX 2000 Telepresence Mode Guidelines

System Level

- The RMX system must be licensed for *Telepresence Mode.*
- The system must be activated with a *Telepresence* enabled license key.

Conference Level

- If the RMX is not licensed for *Telepresence Mode*, the *Telepresence* option is not displayed in the *New Profile* dialog box
- A *Telepresence* conference must have *Telepresence Mode* enabled in its profile.
- When Telepresence mode is selected in a conference profile, the following options are disabled:
  - borders
  - site names
  - speaker indication
  - skins
  - same layout
  - presentation mode
  - auto layout
  - lecture mode
- The master (center) camera is used for video, audio and content.
- *Conference Templates* can be used to simplify the setting up *Telepresence* conferences where precise participant layout and video forcing settings are crucial. *Conference Templates:*
  - Save the conference Profile.
  - Save all participant parameters including their *Personal Layout* and *Video Forcing* settings.
- An ongoing *Telepresence* conference can be saved to a *Conference Template* for later re-use.

  For more information see "*Conference Templates*" on page **8-1**.

Room (Participant/Endpoint) Level

- To the RMX, each camera in a *Telepresence* room is considered to be an endpoint and is configured as a participant.
- The *Telepresence Mode* field is always displayed in the *New Participant* dialog box. If the system is not licensed for *Telepresence* this field is automatically set to None.
- *Telepresence* participants (endpoints) must be specified as:
  - RPX – transmitting 4:3 video

    or
  - TPX – transmitting 16:9 video

RPX and TPX Video Layouts

Additional video layouts have been created to give *Telepresence* operators more video layout options when configuring TPX and RPX room systems. These additional video layout options are only available when *Telepresence* is selected in the conference profile.

***Table 2-9** TPX / RPX – Additional Video Layouts*

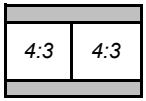
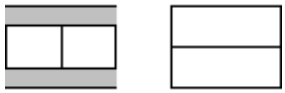
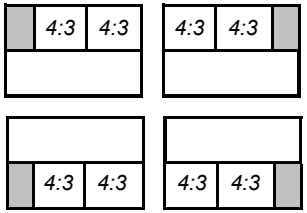
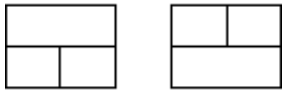
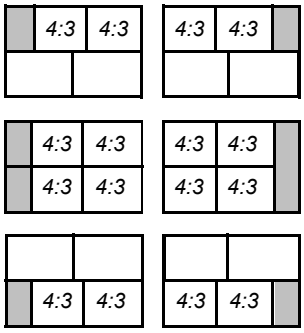
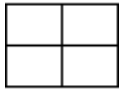
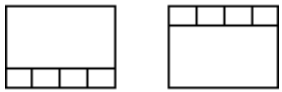
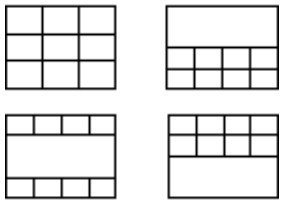
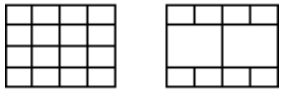
Number of Endpoints	Layouts	
	TPX	RPX
1		
2		
3		
4		

Table 2-9 TPX / RPX – Additional Video Layouts (Continued)

Number of Endpoints	Layouts	
	TPX	RPX
5		
9		
10+		

The following example illustrates the use of standard and additional RMX *Telepresence* layouts when connecting four Room Systems as follows:

- Two TPX Room Systems
  - 2 active cameras
  - 6 screens
- Two RPX Room Systems
  - 8 cameras
  - 8 screens

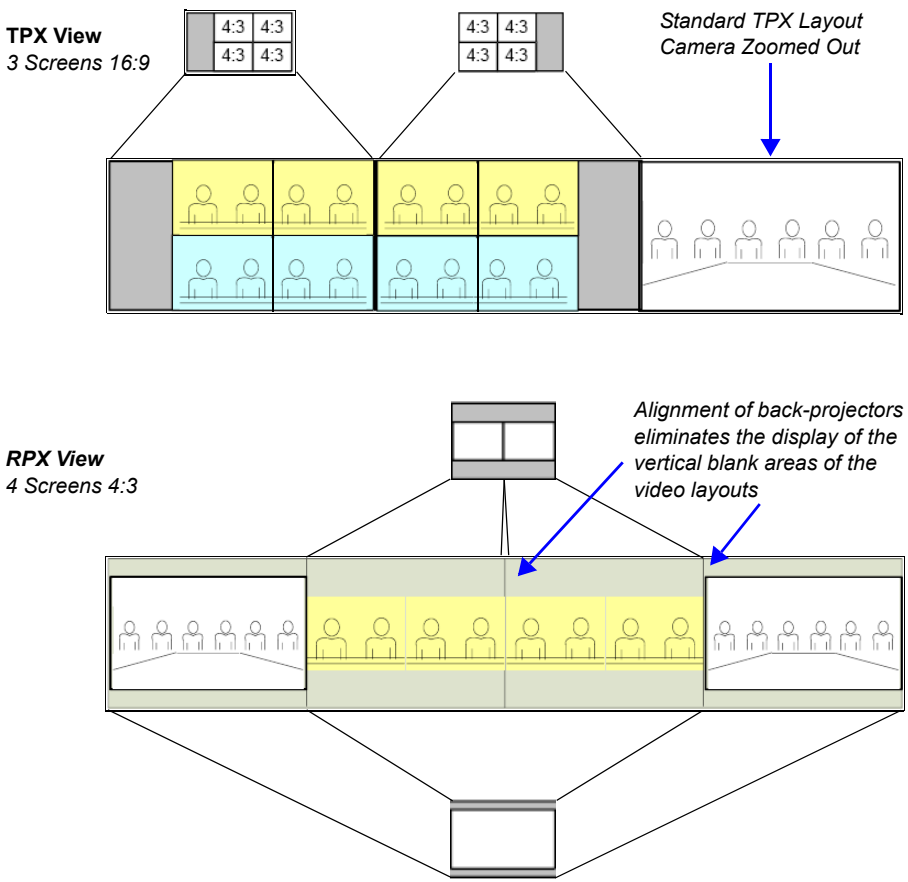


Figure 2-2 *RPX and TPX Room System connected using RMX 2000*

Enabling Telepresence

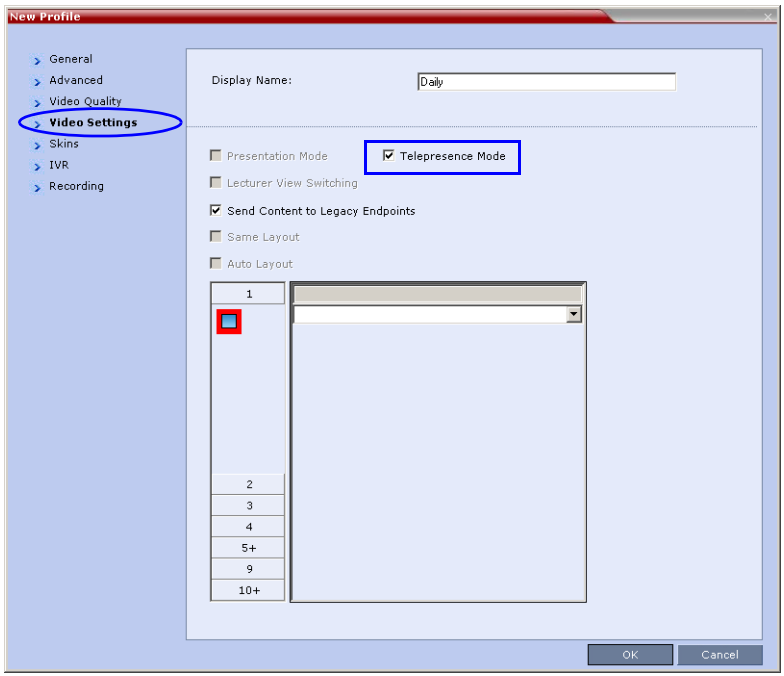
Conference Level

Telepresence Mode must be configured in a new or existing Conference Profile.

To enable Telepresence in a new or existing Conference Profile:

1. In the *RMX Management* pane, click **Conference Profiles**.
2. Click the **New Profiles** ( ) button or open an existing *Conference Profile*.
3. Define the fields of the profile and click the **Video Settings** tab.

   For more information on defining Profiles, see the *RMX 2000 Administrator's Guide*, "*Defining Profiles*" on page **1-8**.
4. Select **Telepresence Mode** to enable the feature in the *Conference Profile*.



5 Select the required video layout.

When Telepresence Mode is enabled, the Skin options are disabled as the system uses a black background and the frames and speaker indication are disabled.

6 Click **OK**.

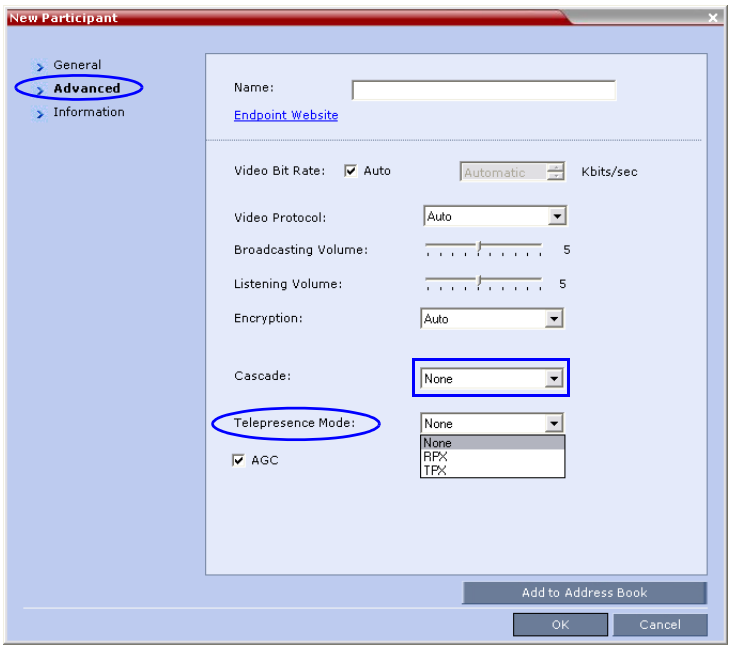
Room (Participant/Endpoint) Level

Setting the participant/endpoint *Telepresence Mode* configures the RMX to receive the video format of the RPX or TPX room endpoints.

To configure a participant/endpoint for Telepresence:

1 In the *Address Book* pane, click **New Participant** ( ) or double-click an existing *Telepresence* endpoint.

The *New Participant* or *Participant Properties - General* dialog box is displayed.



2 If defining a new participant, enter the required information in the *New Participant – General* dialog box for the participant.

For more information, see the *RMX 2000 Administrator's Guide*, "*Adding a new participant to the Address Book*" on page 4-4.

3 Click the **Advanced** tab.

4 Select the *Telepresence Mode* for the participant:

Table 2-10 *New Participant – Telepresence Mode*

Mode	Description
RPX	Select this option for room endpoints that transmit 4:3 video format.
TPX	Select this option for room endpoints that transmit 16:9 video format.
None	Select this option for endpoints that are neither RPX or TPX room endpoints.

5 Click **OK**.

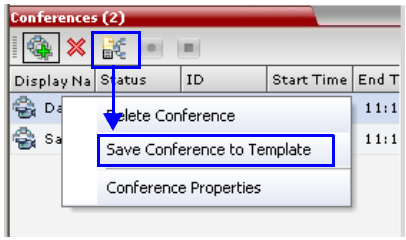
Saving an Ongoing Conference as a Template

Any conference that is ongoing can be saved as a template.

To save an ongoing conference as a template:

1 In the *Conferences List*, select the conference you want to save as a Template.

2 Click the **Save Conference** ( ) button.
or
Right-click and select **Save Conference**.



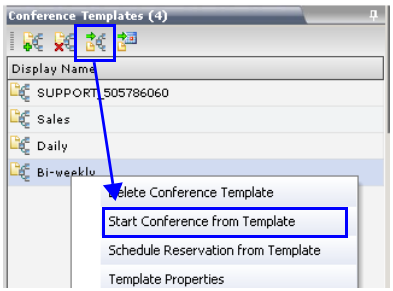
The conference is saved to a template whose name is taken from the ongoing conference *Display Name*.

Starting an Ongoing Conference From a Template

An ongoing conference can be started from any Template saved in the *Conference Templates* list.

To start an ongoing conference from a Template:

1. In the *Conference Templates* list, select the Template you want to start as an ongoing conference.
2. Click the **Start Ongoing Conference** ( ) button.
   or
   Right-click and select **Start Ongoing Conference**.



   The conference is started.

The name of the ongoing conference in the *Conferences* list is taken from the Template display name of the template.

Cascading Conferences - Star Topology

Cascading enables administrators to connect one conference directly to another conference using an H.323 connection, creating one large conference. The conferences can run on the same MCU or different MCUs. There are many reasons for cascading conferences, the most common are:

- Connecting two conferences on different MCUs at different sites.
- Utilizing the connection abilities of different MCUs, for example, different communication protocols, such as, serial connections, ISDN, etc....

The link between the two conferences is created when a participant that is defined as a dial-out cascaded link in one conference (Conference A) connects to the second conference (Conference B) via a special cascaded Entry Queue (EQ). When MCU A dials out to the cascaded link to connect it to conference A, it actually dials out to the cascaded Entry Queue defined on MCU B.

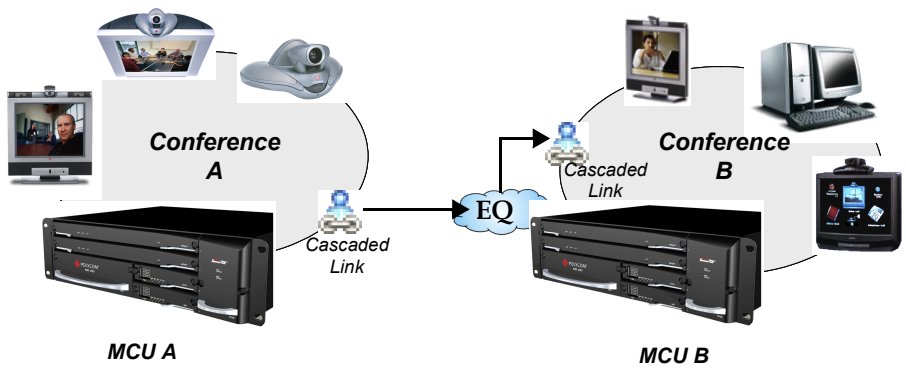


Figure 2-3 Cascaded Conferences - Star Topology

Though the process of cascading conferences mentioned in this section refers to conferences running on two different RMX units, it is possible to cascade conferences running between RMX units and other MCUs.

Simple cascade links are treated as endpoints in CP conferences and are allocated resources according to Table 2-3 on page 2-6. Cascaded links in 1x1 video layout are in SD resolution.

In HD Video Switching, simple cascade links behave like HD endpoints if all HD prerequisites are met - if not, the link is audio only.

When cascading two conferences, the video layout displayed in the cascaded conference is determined by the selected layout in each of the two conferences. Each of the two conferences will inherit the video layout of the other conference in one of their windows.

In order to avoid cluttering in the cascaded window, it is advised to select appropriate video layouts in each conference before cascading them.

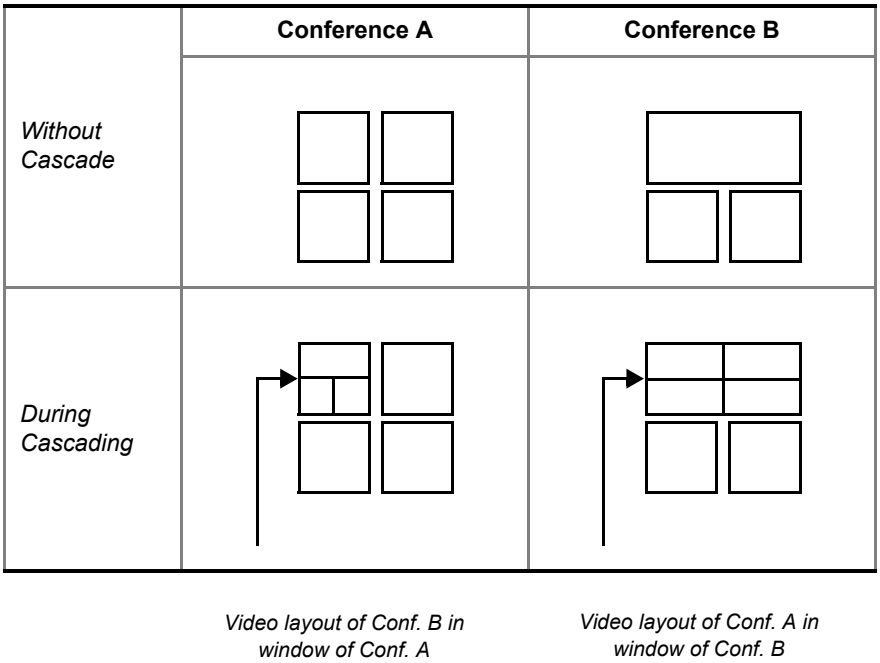

Conference A
Conference B
Without Cascade
During Cascading
Video layout of Conf. B in window of Conf. A
Video layout of Conf. A in window of Conf. B

Figure 2-4 *Video Layouts in Cascaded Conferences*

The following features are not supported by the cascaded link and therefore are not supported in the combined conference:

- **DTMF** codes are enabled in cascaded conference, but only in their local conference. The operations executed via DTMF codes are not forwarded between linked conferences.
- **FECC** (Far End Camera Control will only apply to conferences running in their local MCU).

Enabling Cascading

Cascading two conferences requires that the following procedures are implemented:

- **Creating the cascade-enabled Entry Queue**

  A cascade-enabled Entry Queue must be created in the MCU hosting the destination conference (Conference B). The cascade-enabled Entry Queue is used to establish the dial-in link between the destination conference and the linked conference and bypassing standard Entry Queue, IVR prompt and video slide display.
- **Creating a cascade-enabled Dial-out link**

  The creation of a cascade-enabled dial-out link (participant) in the linked conference (Conference A). This dial-out participant functions as the link between the two conferences.
- (Optional) Enabling the cascaded linked participant to connect to the linked conference (Conference A) without entering the conference password. This can be done by modifying the default settings of the relevant system flag.

Creating the Cascade-enabled Entry Queue

The cascade-enabled Entry Queue maintains the correct behavior of the cascaded link when it dials into it.

The cascade-enabled Entry Queue should be used only to connect cascaded links and should not be used to connect standard participants to conferences.

When cascading High Definition (HD) conferences, the cascade-enabled Entry Queue must have the same settings as both cascaded conferences and the participants in both conferences must use the same line rate and HD capabilities as set for the conferences and Entry Queue.

To define a Cascade-Enabled Entry Queue:

1 In the *RMX Management* pane, click the **Entry Queues** button.

The *Entry Queues* list pane is displayed.

2 Click the **New Entry Queue** button.

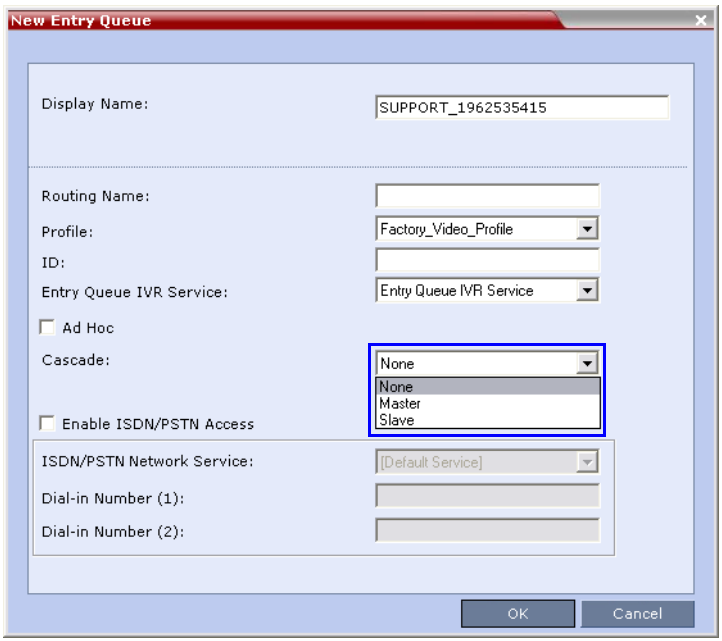
The *New Entry Queue* dialog box is displayed.

3 Define the standard Entry Queue parameters (as described in Chapter 3).

4 In the *Cascade* field, select **Master** or **Slave** depending on the Master/Slave relationship.

— Set this field to **Master** if the Entry Queue is defined on the MCU that is at the center of the topology and other conferences dial into it (acting as the Master).

— Set this field to **Slave** if the Entry Queue is defined on the MCU acting as a Slave, that is, to which the link from the Master MCU (MCU at the center of the topology) is dialing.



If you are defining an HD cascaded Entry Queue, it is recommended to select the same Profile that is selected for both conferences.

5 Click **OK**.

The new Entry Queue enabling cascading is created.

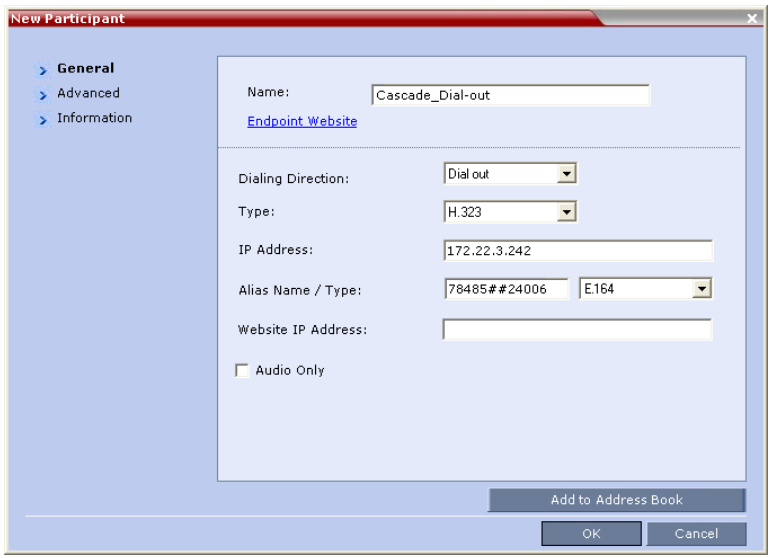
Creating the Dial-out Cascaded Link

The dial-out link (participant) is created or added in the linked conference (Conference A). The dial-out string defined for the participant is the dialing string required to connect to the destination conference (Conference B) Entry Queue defined on the MCU hosting the destination cascaded conference. The dial-out participant can be defined in the Address Book and added to the conference whenever using the same cascade-enabled Entry Queue and a destination conference (with the same ID and Password).

To define the Dial-out Cascaded Link:

1. Display the list of participants in the linked conference (Conference A).
2. In the *Participant List* pane, click the **New Participant** button.

   The *New Participant - General* dialog box is displayed.



3. In the *Name* field, enter a participant name.
4. In the *Dialing Direction* field, select **Dial-out**.
5. In the *Type* list field, verify that **H.323** is selected.

6 There are two methods to define the dialing string:

A Using the MCU's IP Address and the Alias string.

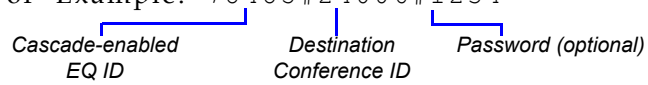
B Using only the Alias string (requires a gatekeeper).

Method A (If no gatekeeper is used):

In the *IP Address* field, enter the IP address of the **Signaling Host** of the MCU hosting the destination conference (in the example, MCU B).

In the *Alias Name/Type* field, enter the ID of the cascade-enabled Entry Queue (EQ), the Conference ID and Password of the destination conference (MCU B) as follows:
EQ ID#Destination Conference ID#Password (Password is optional).

For Example: 78485#24006#1234

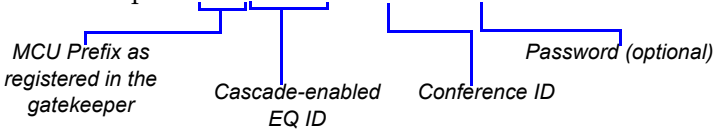


Method B (Using a gatekeeper):

In the *Alias Name* field, enter the Prefix of MCU B, EQ ID, Destination Conference ID, and Password, as follows:

MCU Prefix EQ ID#Conference ID#Password (Password is optional)

For Example: 92578485#24006#1234

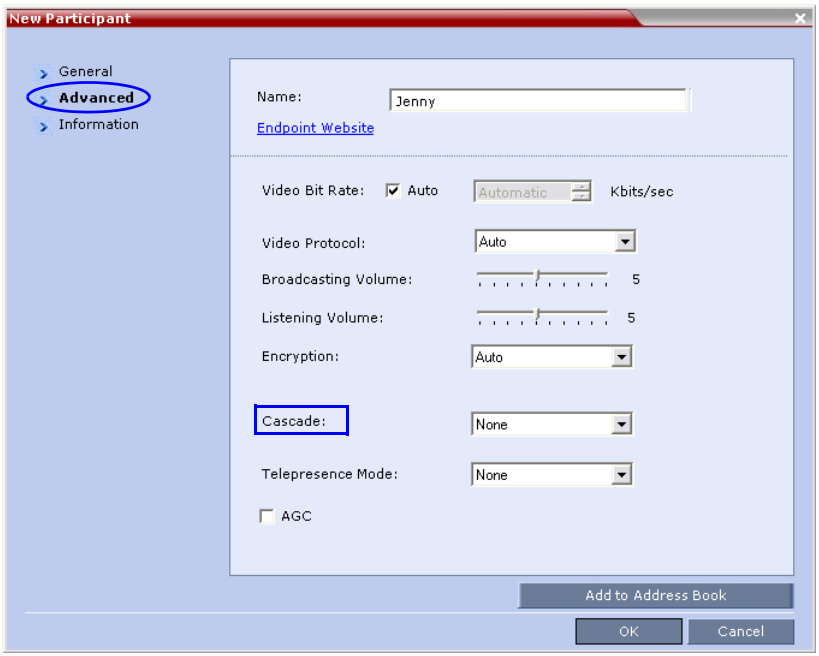


7 Click the **Advanced** tab.

8 In the *Cascade* field, select:

— **Slave**, if the participant is defined in a conference running on a Slave MCU and will connect to the Master MCU (in the center of the topology).

— **Master**, if the participant is defined in a conference running on the Master MCU (in the center of the topology) dialing from the Master MCU to the Slave MCU.



9 Click **OK**.
The cascade-enabled dial-out link is created and the system automatically dials out to connect the participant to the linked conference, as well as the destination conference.

Enabling Cascaded Conferences without Password

If a password is assigned to the linked conference, cascaded links will be prompted for a password when connecting to it (Conference A). Administrators have the option of altering the MCU settings to enable cascaded links to connect without a password.

To enable cascaded links to connect without a password:

1. In the RMX web client connected to MCU A (where the linked conference is running), click **Setup>System Configuration**. The *System Flags* dialog box opens.
2. Set the **ENABLE_CASCADED_LINK_TO_JOIN _WITHOUT_PASSWORD** flag to **YES**.
3. Click **OK**.

For more information, see "*System Configuration*" on page **16-10**.

- Reset the MCU for flag changes to take effect.

Monitoring Cascaded Conferences

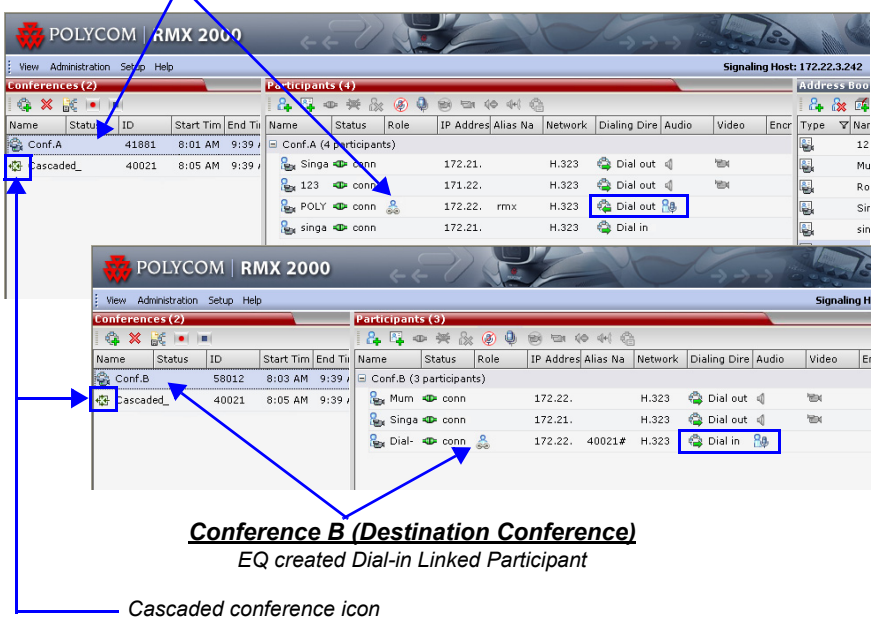
To monitor both conferences at the same time, two instances of the RMX Web Clients must be opened (one for each MCU) by entering the IP Address of each MCU. If both conferences are running on the same MCU, only one RMX Web Client window is required.

When conferences are cascaded, the *Participant* list pane of each of the two conferences will display a linked icon ( ); a dial-in linked icon in the destination conference (Conference B) and a dial-out linked icon in the linked conference (Conference A).

The *Conferences* list panes in each of the two conferences will display a cascaded conference icon ( ) indicating that a conference running on the MCU is presently cascading with another conference running on the same or another MCU. The cascaded conference icon will be displayed for a short period of time and then disappear.

Conference A (Linked Conference)

Dial-out Linked Participant



Conference B (Destination Conference)

EQ created Dial-in Linked Participant

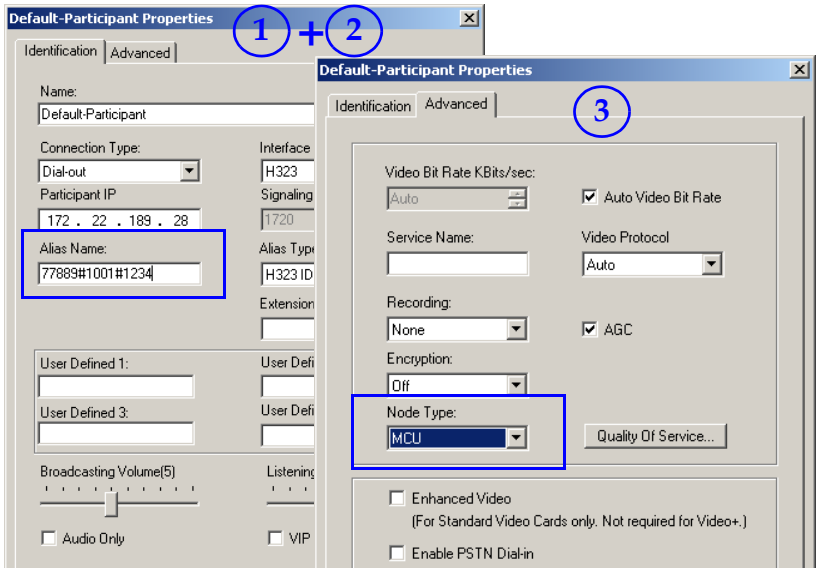
Cascaded conference icon

Creating the Dial-out Link from a Conference Running on the MGC to the Conference Running on the RMX

In the same way that the dial-out cascaded link is created in the RMX, you can create a dial-out participant in the MGC.

In the MGC Manager application, define a new participant as follows:

1. In the *Participant Properties* dialog box, enter a **Participant Name**, select **Dial-out** and **H.323**.
2. Define the **dialing string** as described in step 6 on page 2-54 (both methods are applicable).
3. In the *Advanced* tab's *Node Type* field, select **MCU**.



4. Click **OK**.

Cascading Conferences - H.239-enabled MIH Topology

H.239 Multi-Hierarchy (MIH) cascading is available to RMX users enabling them to run very large conferences on different MCUs in multiple levels of Master-Slave relationships using an H.323 connection.

Multi-Hierarchy (MIH) Cascading is implemented where the cascaded MCUs reside on different networks, whereas *Star Topology Cascading* requires that all cascaded MCUs reside on the same network.

MIH Cascading allows:

- Ability to open and use a content channel (H.239) during conferencing.
- Full management of extremely large, distributed conferences.
- Connecting conferences on different MCUs at different sites.
- Utilizing the connection abilities of different MCUs, for example, different communication protocols, such as, serial connections, ISDN, etc.
- Significant call cost savings to be realized by having participants call local MCUs which in turn call remote MCUs, long distance.

Although participants in MIH Cascading conferences can connect using H.323, SIP and ISDN, the MIH Cascading Links must connect via H.323.

MIH Cascading Levels

The cascading hierarchy topology can extend to four levels (Figure 2-5) and should be deployed according to the following guidelines:

- If an *RMX 2000* is deployed on level 1:
  - — Only *RMX 2000* can be used on level 2, and *DST MCS 4000* and other MCUs can be deployed on levels 3 and 4.
- If an *MGC* is deployed on level 1:
  - — *MGC* or *RMX 2000* can be used on level 2, and *DST MCS 4000* and other MCUs can be deployed on levels 3 and 4.

- *DST MCS 4000 MCU*s connect as endpoints to the *RMX*s or *MGC*s on higher levels.

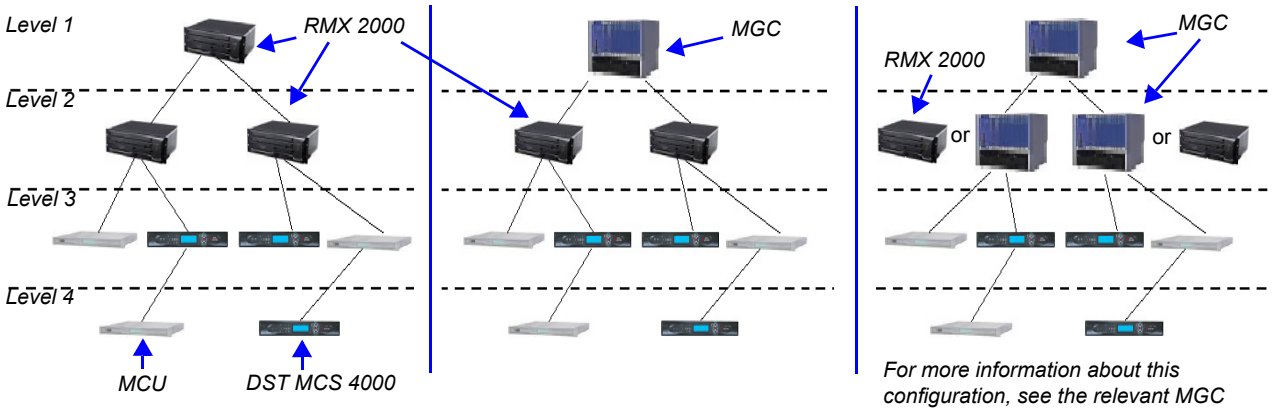


Figure 2-5 *MIH Cascade Levels*

MIH Cascading Guidelines

Master and Slave Conferences

- In *MIH Cascading* conferences, although there are multiple levels of Master and Slave relationships between conferences, the conference that runs on the MCU on level 1 of the hierarchy must be the Master for the entire cascading session. When an MGC is part of the cascading topology, it must be set as Level 1 MCU.
- Conferences running on MCUs on levels 2 and 3 and can be both Masters and Slaves to conferences running on MCUs on levels above and below them.
- All conferences running on MCUs on level 4 are Slave conferences.
- When the DST MCS 4000 is on level 3 and acting as slave to level 2, the RMX 2000 on level 2 must dial out to it in order for the DST MCS 4000 to be identified as slave. The link between the two MCU (dial out participant) is defined as a standard participant and not as a cascading link.

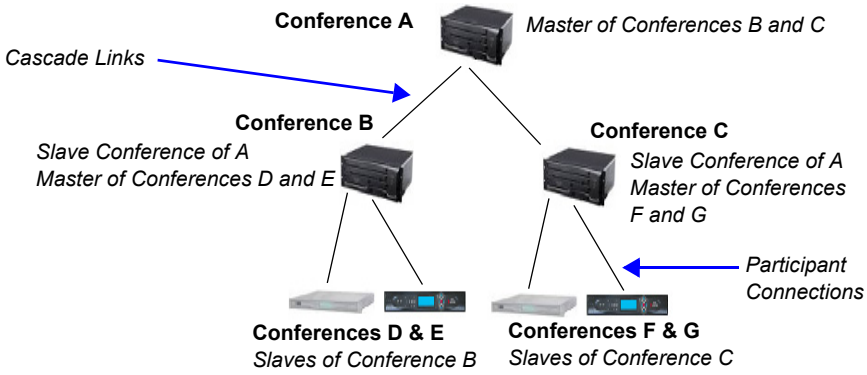


Figure 2-6 MIH Cascading – Master-Slave Relationship

Video Session Mode, Line Rate and Video Settings

The types of MCUs, their position in the cascade topology and the endpoint capabilities (HD/CIF and H.263/H.264) determine the *Video Session Mode* of the *MIH Cascading* conference.

- When creating a cascading link between two RMXs:
  - The RMXs operate in CP (Continuous Presence) mode.
- When creating a cascading link between MGCs and RMXs:
  - If there are no MGCs on level 2, the MGCs can operate in either in CP or VSW (Video Switching) mode.
  - If there are MGCs on level 2, the MGCs can only operate in VSW mode.
- When creating a cascading link between two MGCs:
  - The MGCs must be configured to operate in VSW mode.

  For more details about the MGC to MGC connection, see the *MGC Manager User's Guide, Volume II, Chapter 1, "Ad Hoc Auto Cascading and Cascading Links"*.

To enable the connection of the links between cascaded conferences, they must run at the same line rate.

The following table summarizes *Video Session Modes* line rate options that need to be selected for each conference in the cascading hierarchy according to the cascading topology:

Table 2-11 *MIH Cascading – Video Session Mode and Line Rate*

Topology	MCU Type	Video Session Mode	Line Rate	Endpoint
Level 1	RMX 2000	CP - HD	1.5Mb/s, 1Mb/s, 2Mb/s	HDX
Level 2	RMX 2000			
Level 1	RMX 2000	CP - CIF	768Kb/s, 2Mb/s	VSX
Level 2	RMX 2000			
Level 1	MGC	CP - CIF 263	768Kb/s, 2Mb/s	HDX, VSX
Level 2	RMX 2000	CP - CIF 264		
Level 1	MGC	VSW - HD	1.5Mb/s	HDX
Level 2	RMX 2000	VSW HD		
Level 2	RMX 2000	CP/VSW -HD	1.5Mb/s, 1Mb/s, 2Mb/s	HDX
Level 3	MCS 4000			
Level 2	RMX 2000	CP - CIF	768Kb/s, 2Mb/s	HDX, VSX
Level 3	MCS 4000			

H.239 Content Sharing

Content sharing is controlled by means of a token. The *Content Token* is allocated to participants by the highest level master conference.

- The *Content Token* must be released by the participant that is currently holding it before it can be re-allocated.
- After release, the *Content Token* is allocated to the participant that most recently requested it.
- The *Content Token* can be withdrawn from a conference participant by using the RMX web client only if the highest level master conference is running on the RMX unit.

- The following table lists the bit rate allocated to the Content channel from the video channel in each of the three Content modes:

Table 2-12 Bit Rate Allocation to Content Channel

Conf Kbps / Mode	64/ 96	128	256	384	512	768	1024	1472	1920
Graphics	0	64	64	128	128	256	256	256	256
Hi-res Graphics	0	64	128	192	256	384	384	512	512
Live Video	0	64	128	256	384	512	768	768	768

Setting up MIH Cascading Conferences

The cascading topology, the master/slave relationship and the dialing direction determines the set-up procedure:

- RMX to RMX
- MGC to RMX
- MGC to MGC
  For more details about the MGC to MGC connection, see the *MGC Manager User's Guide, Volume II, Chapter 1, "Ad Hoc Auto Cascading and Cascading Links"*.

RMX to RMX Cascading

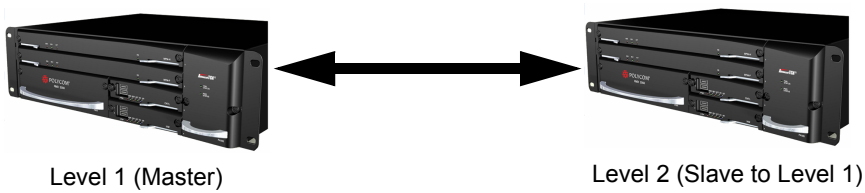


Figure 2-7 Dialing Direction

To establish the links between two RMXs requires the following procedures be performed:

- Establish the Master-Slave relationships between the cascaded conferences by defining the dialing direction.

- Create a cascade-enabled *Entry Queue* for dial-in connections (you create it once for all cascading links using the same line rate).
- Create the Master and Slave conferences, defining the appropriate line rate and whether it is a CP conference or HD Video Switching conference.
- Create a cascade-enabled *Dial-out Participant* link in the Master or the Slave conference (depending on the dialing direction).

Establish the Master-Slave relationships and the dialing direction

MIH Cascading conferences are linked in a master-slave relationship with each other according to the dialing direction. It determines the definition of the cascaded links and the Entry Queues. Dialing directions can be top-down or bottom-up or up from level 4 to level 3 and from level 3 to level 2 and down from level 1 to level 2.

It is recommended to select one dialing direction (usually bottom up) for the entire hierarchy to simplify the setup procedure.

Table 2-13 *Set up Procedures according to the Dialing Direction*

Dialing Direction	RMX 2000 Level 1	RMX 2000 Level 2
RMX 2000 *Level 1* *to* RMX 2000 *Level 2*		Define the cascade-enabled Entry Queue, defining it as **Slave.**
	Define the conference line rate and if required to HD Video Switching to be the same as the one set on the RMX 2000 Level 2.	Define the conference line rate and if required to HD Video Switching to be the same as the one set on the RMX 2000 Level 1.
	Define the dial-out participant (Cascaded Link) to the conference running on the RMX 2000 on Level 2, setting it as **Master.**	

Table 2-13 *Set up Procedures according to the Dialing Direction*

Dialing Direction	RMX 2000 Level 1	RMX 2000 Level 2
RMX 2000 *Level 2* *to* RMX 2000 *Level 1*	Define the cascade-enabled Entry Queue, setting it as **Master**.	
	Define the conference line rate and Video Session Mode to be the same as the one set on RMX Level 2.	Define the conference line rate and Video Session Mode to be the same as the one set on RMX Level 1.
		Define the dial-out participant (Cascaded Link) to the conference running on the RMX 2000 on Level 2, setting it as **Slave.**

- When cascading between a DST MCS 4000 on level 3 and the RMX 2000 on level 2, the RMX 2000 must dial out to the MCS 4000 to establish the Master-Slave relationship (the RMX 2000 is the Master).
- If the RMX 2000 on level 2 is being dialed from both Level 1 and Level 3 and it is acting as both Slave to level 1 and Master to Level 3, two Cascade-enabled Entry Queues must be defined: one defined as Slave (for dial in from conferences running on MCU Level 1) and the other defined as Master (for dial in from conferences running on MCU Level 3).

Creating a Cascade Enabled Entry Queue

Cascade-enabled Entry Queues do not play IVR prompts and video slide displays associated with standard Entry Queues.

Depending on the dialing direction, a cascade-enabled Entry Queue is defined either on the MCU on level 1 or on level 2. (See *Dialing Direction*).

The definition of the Entry Queue as Master or Slave is done accordingly.

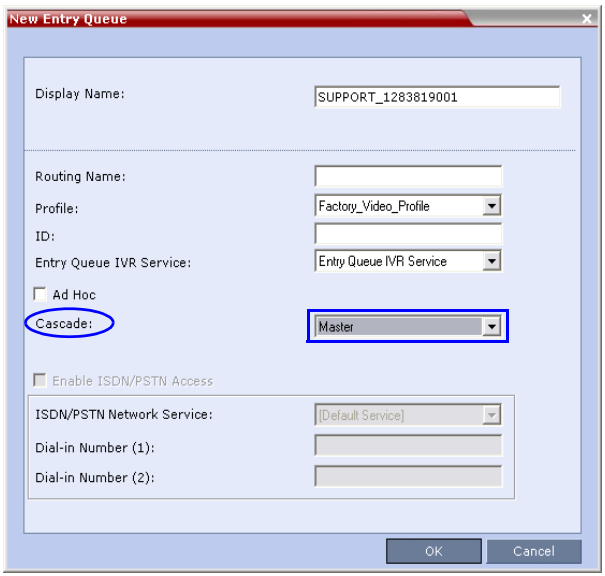
To define a Cascade-Enabled Entry Queue:

1 In the *RMX Management* pane, click **Entry Queues**.

The *Entry Queues* list pane is displayed.

2 Click the **New Entry Queue** ( ) button.

The *New Entry Queue* dialog box is displayed.



3 Define the Entry Queue parameters as for a standard Entry Queue.

For more information about Entry Queue parameters, see the *RMX 2000 Administrator's Guide, Entry Queues* on page **4-1**.

4 In the *Cascade* field, select **Master** or **Slave** depending on the Master/Slave relationship.

— Set this field to **Master** if:

- The Entry Queue is defined on the MCU on level 1 and the dialing is done from level 2 to level 1.
- The Entry Queue is defined on the MCU on level 2 and the dialing is done from level 3 to level 2.

— Set this field to **Slave** if the Entry Queue is defined on the MCU on level 2 (Slave) and the dialing is done from MCU level 1 to level 2.

5 Click **OK**.

Cascade-enabled Entry Queues should not be used to connect standard participants to conferences.

Creating the Cascaded Conferences

The table below lists the line rates that should be used when defining the conference Profiles for cascaded conferences on the RMX 2000 on both Level 1 and Level 2. The video settings will be automatically selected by the system, however, if HD Video Switching is used, it must be selected in the conference Profiles.

Table 2-14 *Recommended Conference Line Rates for Cascaded Conferences*

Topology	Video Session Mode	Conference Line Rate
RMX 2000 ↓ RMX 2000	CP-HD	1.5Mb/s, 1Mb/s, 2Mb/s
	CP-CIF	768Kb/s, 2Mb/s

Creating a Cascade Enabled Dial-out Participant Link

The connection between two cascaded conferences is established by a cascade enabled dial-out participant, acting as a cascades link.

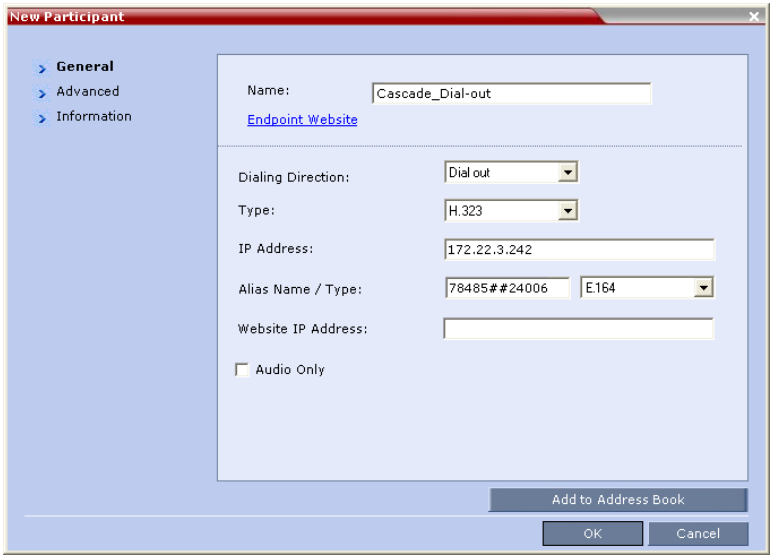
The dialing direction determines whether the dial-out participant is defined in the conference running on the Master MCU or the Slave MCU. For example, if the dialing direction is from level 1 to level 2, and the Master conference is on level 1, the dial-out participant is defined in the conference running on the MCU on level 1 (connecting to an Entry Queue defined as Slave running on the MCU on level 2.

If the cascade-enabled dial-out participant always connects to the same destination conference via the same cascade-enabled Entry Queue on the other (second) MCU, the participant properties can be saved in the Address Book of the MCU for future repeated use of the cascaded link.

To define the dial-out cascade participant link:

1. In the *Conferences* pane, select the conference.
2. In the *Participants* pane, click **New Participant** ( ).

The *New Participant - General* dialog box is displayed.



3 Define the following parameters:

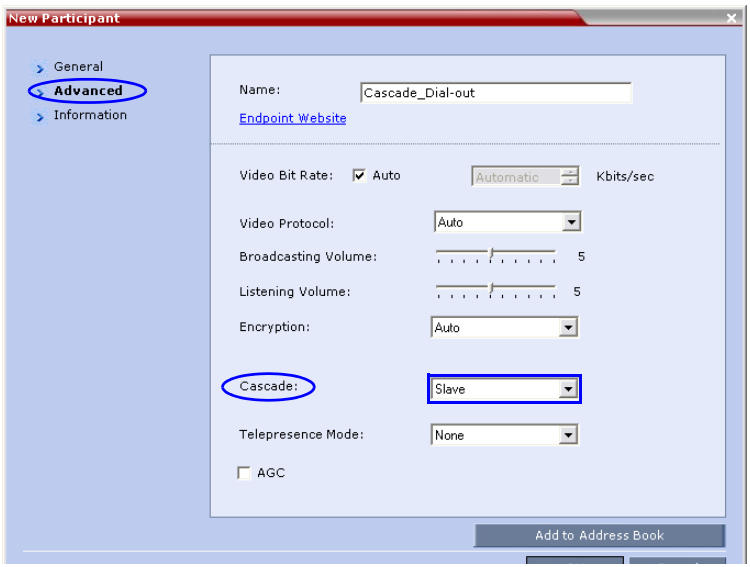
Table 2-15 *New Participant – Dial-out Cascade Link*

Field	Description
Display Name	Enter the participant name
Dialing Direction	Select **Dial-out**.
Type	Select **H.323**.
IP Address	Enter the IP address of the Signaling Host of the MCU running the other (second) conference, where the cascade enabled Entry Queue is defined.

Table 2-15 New Participant – Dial-out Cascade Link (Continued)

Field	Description
Alias Name	If you are using the target MCU IP address, enter the dial string made up of the ID of the cascade enabled Entry Queue and the Conference ID as follows: <Cascade_Enabled_Entry_Queue_ID> ## <Conference_ID> For example: 78485##24006 If a gatekeeper is used, you can enter the prefix of the target MCU, registered with the gatekeeper, instead of the IP address, as part of the dialing string. **<Gatekeeper_Prefix>**<Cascade_Enable_ Entry_ Queue_ID>##<Conference_ID> For example: **925**78485##24006 If the conference has a password and you want to include the password in the dial string, append the password to in the dial string after the Conference ID. For example: 78485##24006**##1234** If the conference has a password and you do not want to include the password in the dial string, set the ENABLE_CASCADED_LINK_TO_JOIN_WITHOUT_PASSWORD flag to **YES**. For more information see the *RMX2000 Administrator's Guide, "Modifying System Flags"* on page **11-5**.
Alias Type	Select **E.164** (digits 0-9, *, #).

4 Click the *Advanced* tab.



5 In the *Cascade* field, select:

— **Slave**, if the participant is defined in a conference running on a Slave MCU.

— **Master**, if the participant is defined in a conference running on the Master MCU.

6 Click **OK**.

MGC to RMX 2000 Cascading

Level 1 (Master) Level 2 (Slave to Level 1)

MGC is always on level 1 and must be set as the Master MCU. If the cascading topology includes additional MGCs as well as RMXs it is recommended to define Video Switching conferences for all the cascading conferences in the topology.

Depending on the dialing direction, the following procedures must be performed:

Table 2-16 *Set up Procedures according to the Dialing Direction*

Dialing Direction	MGC Level 1	RMX 2000 Level 2
MGC to RMX 2000	Set the appropriate flags (done once only).	Set the appropriate flags (done once only).
		Define the cascade-enabled Entry Queue, setting it as **Slave**.
	Define the conference setting and its line rate to be the same as the one set on the RMX 2000.	Define the conference setting and its line rate to be the same as the one set on the MGC.
	Define the dial-out participant (Cascaded Link) to the conference running on the RMX 2000.	

Table 2-16 Set up Procedures according to the Dialing Direction

Dialing Direction	MGC Level 1	RMX 2000 Level 2
RMX 2000 to MGC	Set the appropriate flags (done once only)	Set the appropriate flags (done once only)
	Define the cascade-enabled Entry Queue.	
	Define the conference setting and its line rate to be the same as the one set on the RMX 2000.	Define the conference setting and its line rate to be the same as the one set on the MGC.
		Define the dial-out participant (Cascaded Link) to the conference running on the MGC, setting the participant Cascade parameter to **Slave**.

Setting the flags in the MGC

Flag setting is required to ensure the correct MCU behavior for cascading conferences. It is performed once per MCU.

1 In the MGC Manager, right-click the *MCU icon* and then click **MCU Utils>Edit "system.cfg"**.

2 In the **H264 Section**, ensure that the following flags are set to:

— **ENABLE_HD_SD_IN_FIXED_MODE=YES**
Setting this flag to YES enables H.264 Standard Definition (SD), High Definition (HD) and VSX 8000 (Version 8.0) support in Video Switching conferences.

— **H264_VSW_AUTO=NO**
Setting this flag to NO disables the highest common mechanism in H.264 and enables the selection of H.264 Video Protocol in

fixed mode in Dual Stream Video Switching cascading conferences

— **ENABLE_H239_ANNEX_T=YES**
This flag should be set to the same value (YES/NO) as the settings of the RMX flag H263_ANNEX_T

To use MIH Cascade in the MGC, the Conference Numeric ID routing mode must be used. It is determined when the system.cfg flag in the GREET AND GUIDE/ IVR section is set to QUICK_LOGIN_VIA_ENTRY_QUEUE=NO.

3 Click **OK**.

4 If you changed the flags, reset the MCU.

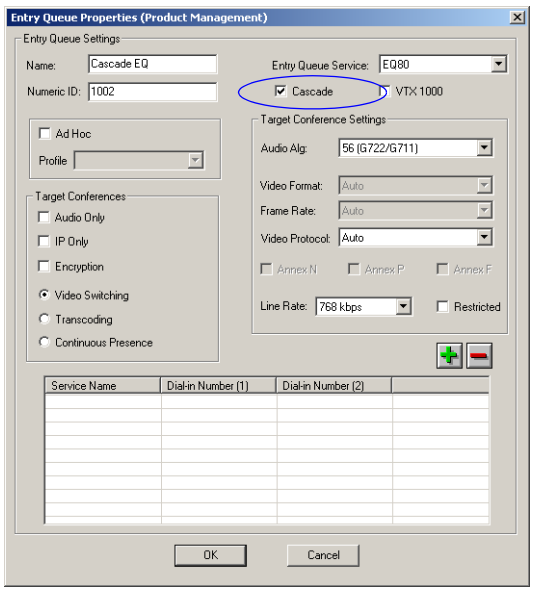
Defining the Cascading Entry Queue in the MGC

The Entry Queue definition on the MGC is required if the dialing is done from the RMX 2000 to the MGC.

1 In the MGC Manager, expand the *MCU tree*.

2 Right-click the *Meeting Rooms, Entry Queues and SIP Factories* icon and click **New Entry Queue**.

3 In the *New Entry Queue* dialog box, set the Entry Queue parameters and select the **Cascade** check box.



For more details on the definition of new Entry Queues refer to the *MGC Manager User's Guide, Volume II, Chapter 1, "Ad Hoc Auto Cascading and Cascading Links"*.

4 Click **OK**.

Creating the Dial-out Link between the Conference Running on the MGC and the Conference Running on the RMX

If the dialing is done from the MGC to the RMX, you need to define the cascaded link (dial-out participant) in the conference running on the MGC.

The dial-out string defined for the participant is the dialing string required to connect to the destination conference via the Cascade-enabled Entry Queue defined on the RMX hosting the destination cascaded conference. The dial-out participant can be defined on the MGC as template or assigned to the Meeting Room.

In the MGC Manager application, define a new participant as follows:

1 In the *Participant Properties - Identification* dialog box, enter a **Participant Name**

2 In the *Connection Type* field, select **Dial-out**.

3 In the *Interface Type* list field, select **H.323**.

4 There are two methods to define the dialing string to the other conference:

   a Using the MCU's IP Address and the Alias string.

   b Using only the Alias string (requires a gatekeeper).

   Method A (If no gatekeeper is used):

   In the *IP Address* field, enter the IP address of the **Signaling Host** of the RMX 2000 hosting the destination conference.

   In the *Alias Name/Type* field, enter the ID of the cascade-enabled Entry Queue (EQ), the Conference ID and Password of the destination conference as follows:

   EQ ID##Destination Conference ID##Password (Password is optional).

   For Example: 1002##12001##1234

   Cascade-enabled EQ ID — *Destination Conference ID* — *Password (optional)*

Method B (Using a gatekeeper):

In the *Alias Name* field, enter the Prefix of MCU B, EQ ID, Destination Conference ID, and Password, as follows:

MCU Prefix EQ ID##Conference ID##Password (Password is optional)

For Example: 9251002##12001##1234

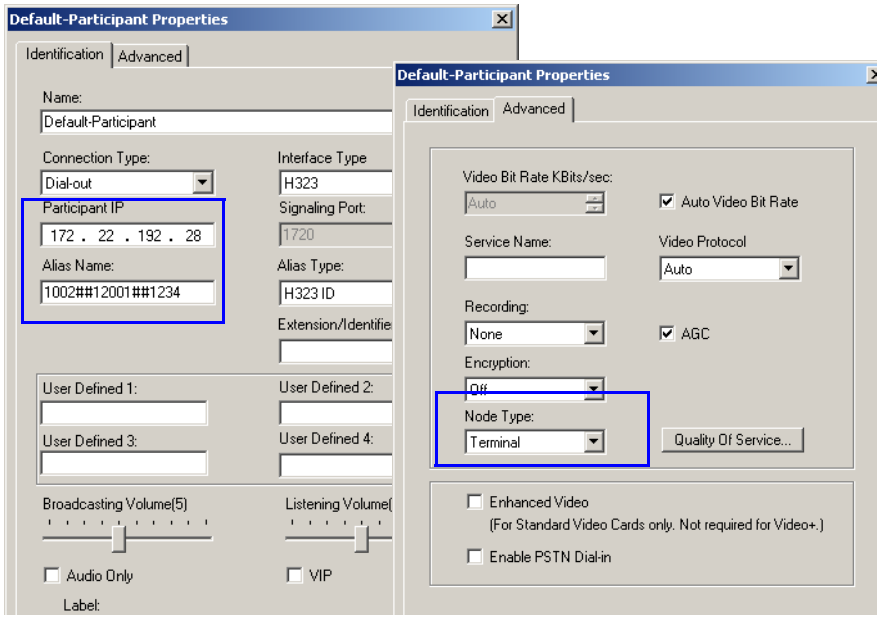
MCU Prefix as registered in the gatekeeper

Cascade-enabled EQ ID

Conference ID

Password (optional)

5 Click the *Advanced* tab and in the *Node Type* field, select **Terminal**.



6 Click **OK**.

Setting the Flags on the RMX 2000

When running conferences in mixed environment (RMX 2000 and MGC) there may be small differences between the line rates each MCU is sending. In the RMX 2000, several flags must be set to ensure that these differences will not cause the cascaded link to connect as Secondary and that Content flows correctly between the cascaded conferences. This procedure is performed once per RMX.

1 In the RMX Web Client menu, click **Setup>System Configuration**.

2 In the *System Flags* dialog box, add the following new flags and values:

— **MIX_LINK_ENVIRONMENT=YES**
Setting this flag to YES will adjust the line rate of HD Video Switching conferences run on the RMX 2000 from 1920Kbps to 17897Kbps to match the actual rate of the HD Video Switching conference running on the MGC. In such case, the conference can include IP and ISDN participants.

— **IP_ ENVIRONMENT_LINK=NO**
Setting this flag to YES will adjust the line rate of HD Video Switching conferences run on the RMX 2000 from 1920Kbps to 18432Kbps to match the actual rate of the IP Only HD Video Switching conference running on the MGC. In such case, the conference can include IP Only participants.

If the flag MIX_LINK_ENVIRONMENT is set to YES, the *IP_LINK_ENVIRONMENT* flag must be set to NO.
If the flag MIX_LINK_ENVIRONMENT is set to NO, the *IP_LINK_ENVIRONMENT* flag must be set to YES.

— **H263_ANNEX_T=YES (default)**
This flag enables/disables the use of Annex T with H263. Set it to NO if the endpoints connecting to the conference do not support this mode. In such a case, you must also change the MGC flag ENABLE_H239_ANNEX_T setting to NO.

— **FORCE_1X1_LAYOUT_ON_CASCADED_LINK _CONNECTION=YES** (default).
Set this flag to NO If the MGC is functioning as a Gateway and participant layouts on the other network are not to be forced to 1X1.

3 If the MGC is dialing the RMX and the cascaded link connects to the conference via the Cascade-enabled Entry Queue without being prompted for the conference password, set the flag to YES as follows:

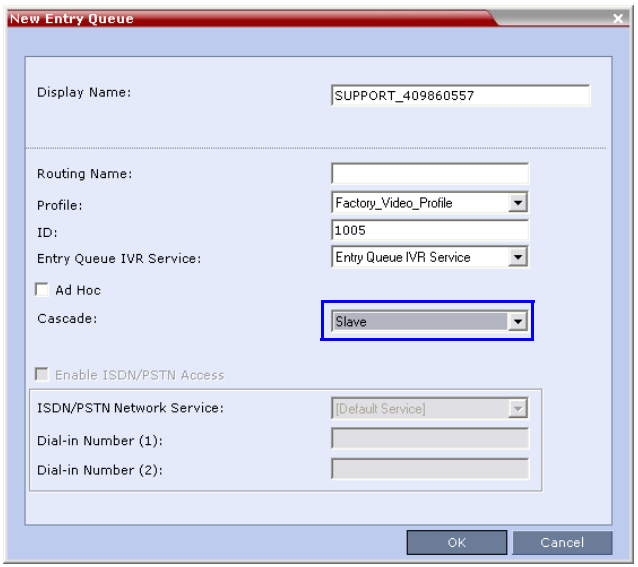
— **ENABLE_CASCADED_LINK_TO_JOIN_WITHOUT_PASSWORD=YES**

4 Click **OK**.

5 Reset the MCU to apply the changes.

Defining the Cascade Enabled Entry Queue on the RMX 2000

If the dialing is done from the conference running on the MGC that is the Master MCU, a Cascade-enabled Entry Queue must be defined on the RMX 2000 setting it as **Slave**.



For more details, see the RMX 2000 to RMX 2000 Cascading.

Defining the Cascading Conferences

The table below lists the line rates and the video settings that should be used when defining the conferences on the MGC. The same line rates should be selected when defining the Conference Profiles on the RMX 2000, as well as whether the conference is HD Video Switching. However, the video settings will be automatically selected by the system.

Table 2-17 *Recommended Conference Line Rates for Cascaded Conferences*

Topology	Video Session Mode	Conference Line Rate
MGC ↓ RMX 2000	MGC - CIF 263 RMX2000 - CIF 264 CP	768Kb/s, 2Mb/s
	MGC - HD VSW RMX2000 - HD VSW	1.5Mb/s

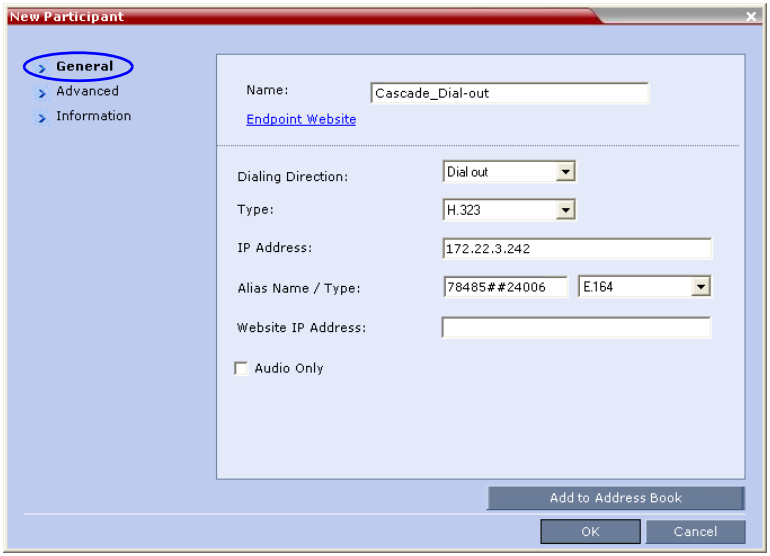
In addition, the conference running on the MGC should be set as **Meet Me Per Conference** and select the **H.239** option in the *Dual Stream Mode* field. *For more details on conference definition on the MGC, refer to the MGC Manager User's Guide, Volume I, Chapter 5*.

Defining the Dial-out Participant on the RMX 2000

If the dialing is done from a conference running on the RMX 2000 to the conference running on the MGC, the dial-out participant is defined in the conference running on the RMX, setting the *Cascade* field to **Slave**. This participant dials the Cascade-enabled Entry Queue defined on the MGC.

1. Display the list of participants in the linked conference (Slave conference).
2. In the *Participant List* pane, click the **New Participant** ( ) button.

The *New Participant - General* dialog box is displayed.



3 In the *Name* field, enter a participant name.

4 In the *Dialing Direction* field, select **Dial-out**.

5 In the *Type* list field, verify that **H.323** is selected.

6 There are two methods to define the dialing string:

A Using the MCU's IP Address and the Alias string.

B Using only the Alias string (requires a gatekeeper).

Method A (If no gatekeeper is used):

In the *IP Address* field, enter the IP address of the MGC hosting the destination conference (Master conference).

In the *Alias Name/Type* field, enter the ID of the cascade-enabled Entry Queue (EQ), the Conference ID and Password of the destination conference (Master Conference) as follows:
`EQ ID##Destination Conference ID##Password` (Password is optional).

For Example: `1005##20006##1234`

Cascade-enabled EQ ID — *Destination Conference ID* — *Password (optional)*

Method B (Using a gatekeeper):

In the *Alias Name* field, enter the MGC Prefix as registered in the gatekeeper, EQ ID, Destination Conference ID, and Password, as follows:

`MGC Prefix EQ ID##Conference ID##Password` (Password is optional)

For Example: `9251005##20006##1234`

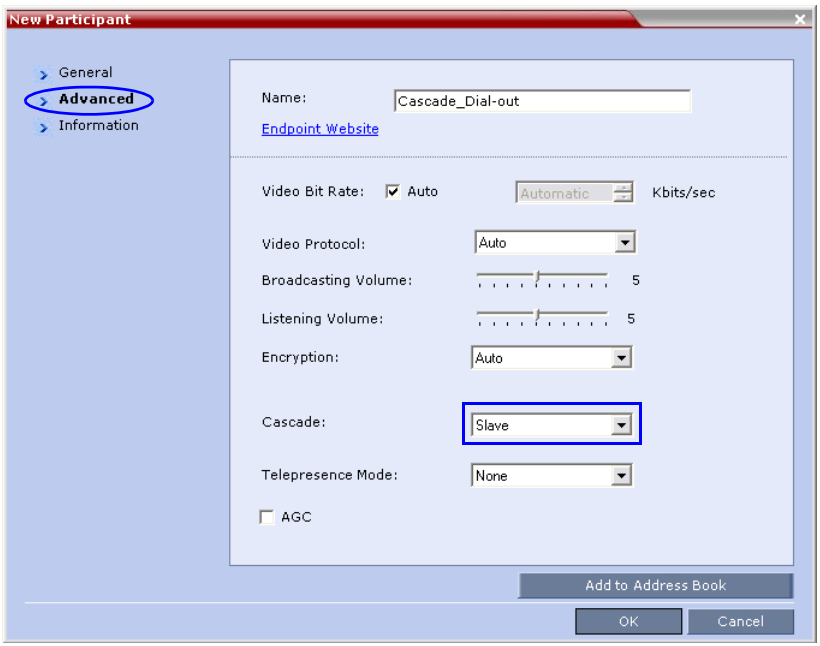
MCU Prefix as registered in the gatekeeper

Cascade-enabled EQ ID

Conference ID

Password (optional)

7 Click the *Advanced* tab and in the *Cascade* field, select the **Slave** option.



8 Click **OK**.

The cascade-enabled dial-out link is created and the system automatically dials out to connect the participant to the local conference, as well as the destination conference on the MGC.

Starting and Monitoring MIH Cascading Conferences

MIH cascading conferences are started in the same way as standard conferences.

- Cascade enabled dial-out link participants on RMX 2000 MCUs are connected automatically.
- Cascade enabled dial-out link participants on MGC MCUs must be connected manually.

For more information on connecting cascade enabled dial-out participant links on other MCU's, refer to their respective operating manuals.

Monitoring Participants in an MIH Cascaded Conference

Once connection between two or more conferences is established, *RMX Web Client* users are able to monitor the following:

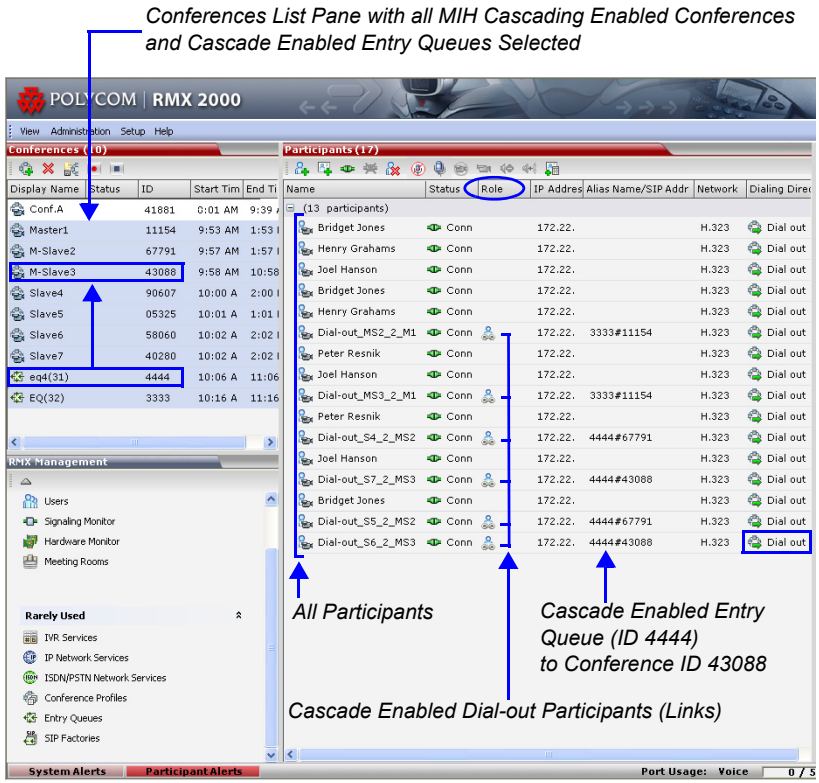
- Master and slave conferences
- Active cascade enabled entry queues – designated with an icon ( ) in the *Role* field of the *Participants List*.
- Cascade enabled dial-out participants (links) – designated with an icon ( ) in the *Display Name* field of the *Conferences List*.

  This indicator is displayed during the connection process and is then removed from the *Conferences List*.

To monitor cascading enabled conferences:

- In the *Conferences List* pane, select all the *MIH Cascading* enabled conferences.

  All the *MIH Cascading* conference participants are displayed:

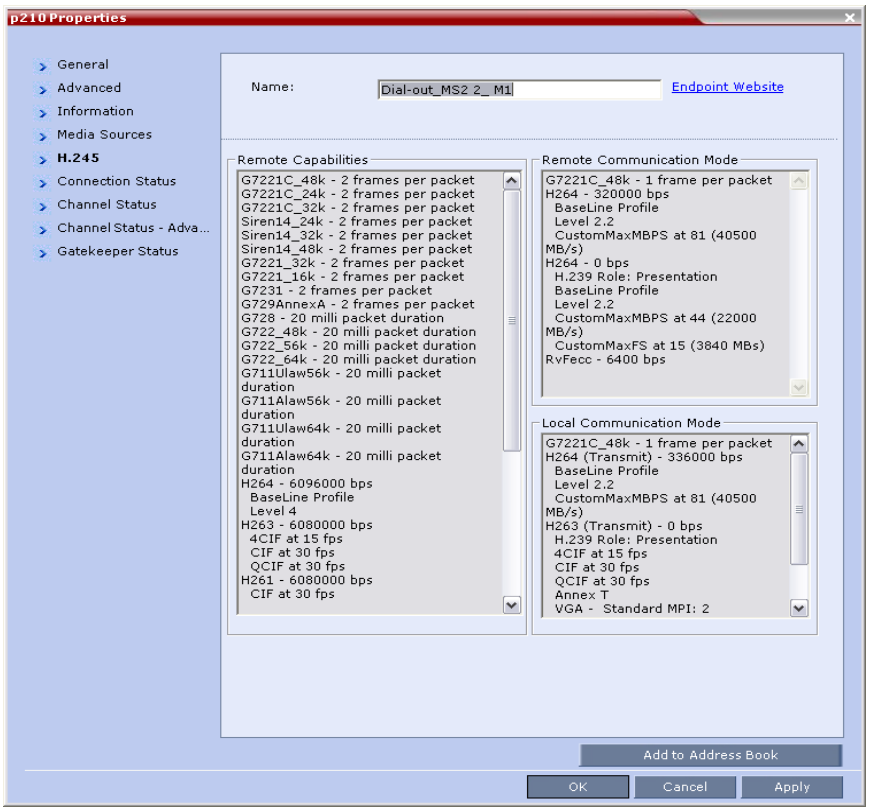

Conferences List Pane with all MIH Cascading Enabled Conferences and Cascade Enabled Entry Queues Selected
All Participants
Cascade Enabled Entry Queue (ID 4444) to Conference ID 43088
Cascade Enabled Dial-out Participants (Links)

Viewing Participant Properties

Viewing *Participant Properties* enables *RMX Web Client* users to view the connection capabilities and status of the link.

To view the linked Participant Properties:

- In the *Participants List* pane, double-click or right-click and select **Participant Properties** of the desired Dial-out linked participant. The Participant Properties dialog box is displayed.



For more information see the *RMX 2000 Administrator's Guide*, "*Participant Level Monitoring*" on page **9-14**.

3

Meeting Rooms

A Meeting Room is a conference saved on the MCU in passive mode, without using any of the system resources. A Meeting Room is automatically activated when the first participant dials into it.

ISDN/PSTN participants can dial-in directly to a Meeting Room without connection through an Entry Queue. Up to two numbers can be defined per conference provided that they are from the same *ISDN/PSTN Network Service*. When a dial-in number is allocated to a Meeting Room, the number cannot be deleted nor can the *ISDN/PSTN Network Service* be removed. The dial-in number must be communicated to the ISDN or PSTN dial-in participants.

Dial-out participants can be connected to the conference automatically, or manually. In the automatic mode the system calls all the participants one after the other. In the manual mode, the RMX user or meeting organizer instructs the conferencing system to call the participant. Dial-out participants must be defined (mainly their name and telephone number) and added to the conference. This mode can only be selected at the conference/Meeting Room definition stage and cannot be changed once the conference is ongoing.

Meeting Rooms can be activated as many times as required. Once activated, a Meeting Room functions as any ongoing conference.

All Meeting Rooms are based on a Profile.

A maximum of 1000 Meeting Rooms can be defined in the system.

The system is shipped with four default Meeting Rooms as shown in Table 3-1.

Table 3-1 *Default Meeting Rooms List*

Meeting Room Name	ID	Default Line Rate
Maple_Room	1001	384 Kbps

Table 3-1 *Default Meeting Rooms List*

Meeting Room Name	ID	Default Line Rate
Oak_Room	1002	384 Kbps
Juniper_Room	1003	384 Kbps
Fig_Room	1004	384 Kbps

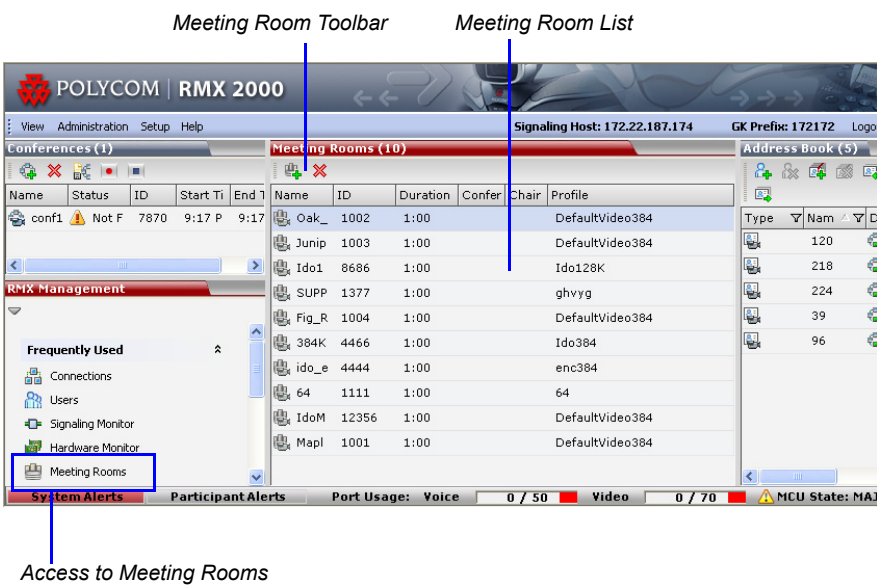
Meeting Rooms List

Meeting Rooms are listed in the *Meeting Room* list pane.

To list Meeting Rooms:

1 In the *RMX Management* pane, in the *Frequently Used* list, click the **Meeting Rooms** button .

The *Meeting Rooms List* is displayed.



An active Meeting Room becomes an ongoing conference and is monitored in the same way as any other conference.

The *Meeting Room List* columns include:

Table 3-2 *Meeting Rooms List Columns*

Field	Description	
Display Name	Displays the name and the icon of the Meeting Room in the *RMX Web Client*.	
	(green)	An active video Meeting Room that was activated when the first participant connected to it.
	(gray)	A passive video Meeting Room that is waiting to be activated.
Routing Name	The ASCII name that registers conferences, Meeting Rooms, Entry Queues and SIP Factories in the various gatekeepers and SIP Servers. In addition, the Routing Name is also: • The name that endpoints use to connect to conferences. • The name used by all conferencing devices to connect to conferences that must be registered with the gatekeeper and SIP Servers.	
ID	Displays the Meeting Room ID. This number must be communicated to H.323 conference participants to enable them to dial in.	
Duration	Displays the duration of the Meeting Room in hours using the format HH:MM (default 01:00).	
Conference Password	The password to be used by participants to access the Meeting Room. If blank, no password is assigned to the conference. This password is valid only in conferences that are configured to prompt for a conference password in the IVR Service.	
Chairperson Password	Displays the password to be used by the users to identify themselves as *Chairpersons.* They are granted additional privileges. If left blank, no chairperson password is assigned to the conference. This password is valid only in conferences that are configured to prompt for a chairperson password.	
Profile	Displays the name of the Profile assigned to the Meeting Room. For more information, see the *RMX Administrator's Guide*, "*Conference Profiles"* on page **1-1**.	

Meeting Room Toolbar & Right-click Menu

The Meeting Room toolbar and right-click menus provide the following functionality:

Table 3-3 Meeting Room Toolbar and Right-click Menus

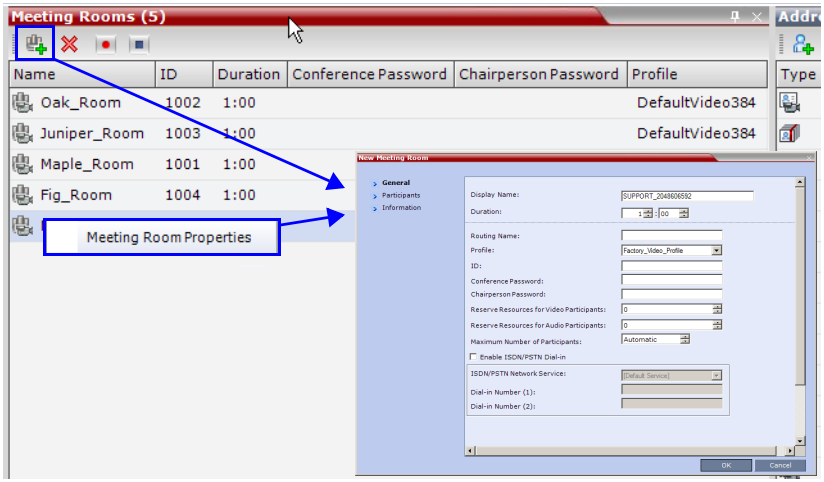
Toolbar button	Right-click menu	Description
	New Meeting Room	Select this button to create a new Meeting Room.
	Delete Meeting Room	Select any Meeting Room and then click this button to delete the Meeting Room.

Dial out to participants assigned to a Meeting Room will only start when the dial in participant who has activated it has completed the connection process and the Meeting Room has become an ongoing conference.

Creating a New Meeting Room

- In the *Meeting Rooms* pane, click the **New Meeting Room** button *or* right-click an empty area in the pane and then click **New Meeting Room**.

  The *New Meeting Room* dialog box appears.



The definition procedure is the same as for the new conference (with the exception of *Reserved Resources* for *Audio* and *Video* participants).

For more information, see the *RMX 2000 Getting Started Guide*, "*Starting a Conference*" on page **3-15**.

4

Entry Queues, Ad Hoc Conferences and SIP Factories

Entry Queues

An Entry Queue (EQ) is a special routing lobby to access conferences. Participants connect to a single-dial lobby and are routed to their destination conference according to the Conference ID they enter. The Entry Queue remains in a passive state when there are no callers in the queue (in between connections) and is automatically activated once a caller dials its dial-in number. The connection of ISDN/PSTN participants to conferences is enabled only via Entry Queues to which an ISDN/PSTN dial-in number is assigned.

A maximum of 40 Entry Queues can be defined in the system.

The parameters (bit rate and video properties) with which the participants connect to the Entry Queue and later to their destination conference are defined in the Conference Profile that is assigned to the Entry Queue. For example, if the Profile Bit Rate is set to 384 Kbps, all endpoints connect to the Entry Queue and later to their destination conference using this bit rate even if they are capable of connecting at higher bit rates.

An *Entry Queue IVR Service* must be assigned to the Entry Queue to enable the voice prompts guiding the participants through the connection process. The Entry Queue IVR Service also includes a video slide that is displayed to the participants while staying in the Entry Queue (during their connection process).

Different Entry Queues can be created to accommodate different conferencing parameters (by assigning different Profiles) and prompts in different languages (by assigning different *Entry Queue IVR Services*). For more information, see "*IVR Services*" on page **13-1**.

The Entry Queue can also be used for Ad Hoc conferencing. If the Ad Hoc option is enabled for the Entry Queue, when the participant enters the target conference ID the system checks whether a conference with that ID is already running on the MCU. If not, the system automatically creates a new ongoing conference with that ID. For more information about Ad Hoc conferencing, see "*Ad Hoc Conferencing*" on page **4-10**.

An Entry Queue can be designated as Transit Entry Queue to which calls with dial strings containing incomplete or incorrect conference routing information are transferred. For more information, see "*Transit Entry Queue*" on page **4-8**.

To enable ISDN/PSTN participants to dial in to the Entry Queue, an ISDN/PSTN dial-in number must be assigned to the Entry Queue. Up to two dial-in numbers can be assigned to each Entry Queue. The dial-in numbers must be allocated from the dial-in number range defined in the ISDN/PSTN Network Service. You can allocate the two dial-in numbers from the same ISDN/PSTN Network Service or from two different ISDN/PSTN Network Services. The dial-in number must be communicated to the ISDN or PSTN dial-in participants.

The Entry Queue can also be used as part of the Gateway to Polycom® Distributed Media Application™ (DMA™) 7000 solution for connecting Audio only PSTN, ISDN, SIP and H.323 endpoints to DMA™ 7000.

For more information, see Appendix H, "*Gateway to Polycom® DMA™ 7000*".

Default Entry Queue properties

The system is shipped with a default Entry Queue whose properties are:

Table 4-1 *Default Entry Queue Properties*

Parameter	Value
Display Name	DefaultEQ The user can change the name if required.
Routing Name	DefaultEQ The default *Routing Name* cannot be changed.

Table 4-1 Default Entry Queue Properties (Continued)

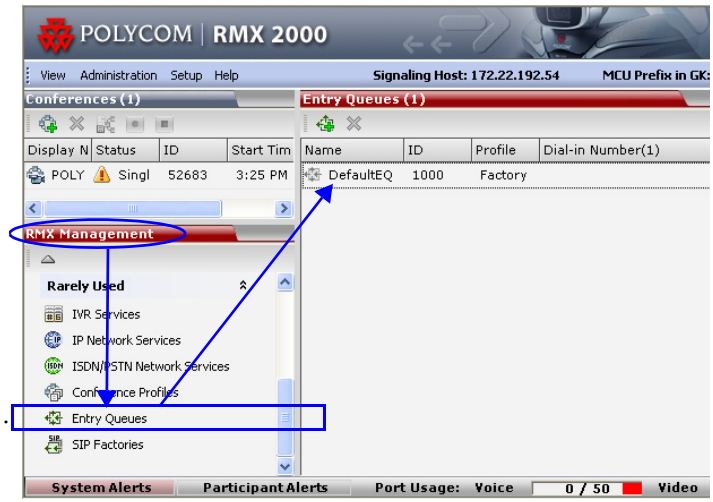
Parameter	Value
ID	1000
Profile name	Factory-Video-Profile. Profile Bit Rate is set to 384 Kbps.
Entry Queue Service	Entry Queue IVR Service. This is default Entry Queue IVR Service shipped with the system and includes default voice messages and prompts in English.
Ad Hoc	Enabled
Cascade	None (Disabled)
Enable ISDN/ PSTN Access	Disabled. You can modify the properties of this Entry Queue to enable ISDN/PSTN participants to dial-in to a conference. Up to two dial-in numbers can be assigned.

Defining a New Entry Queue

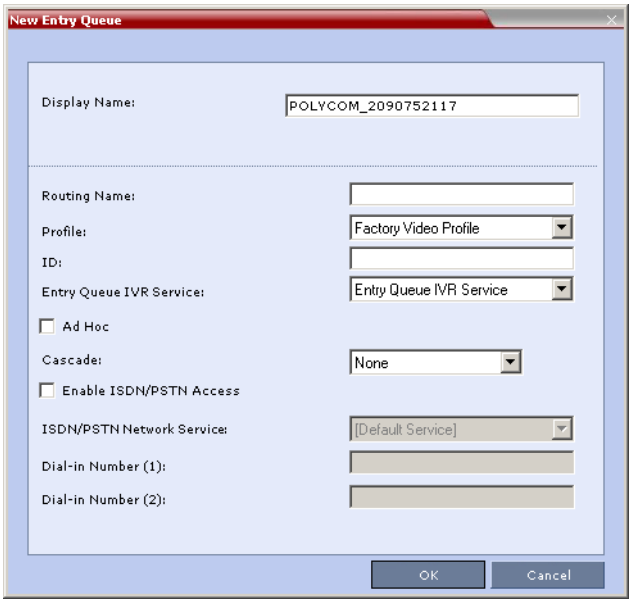
You can modify the properties of the default Entry Queue and define additional Entry Queues to suit different conferencing requirements.

To define a new Entry Queue:

1 In the *RMX Management - Rarely Used* pane, click **Entry Queues.**



2 In the *Entry Queues* list pane, click the **New Entry Queue** button. The *New Entry Queue* dialog box opens.



3 Define the following parameters:

Table 4-2: Entry Queue Definitions Parameters

Option	Description
Display Name	The Display Name is the conferencing entity name in native language character sets to be displayed in the RMX Web Client. In conferences, Meeting Rooms, Entry Queues and SIP factories the system automatically generates an ASCII name for the *Display Name* field that can be modified using Unicode encoding. • English text uses ASCII encoding and can contain the most characters (length varies according to the field).

Table 4-2: Entry Queue Definitions Parameters (Continued)

Option	Description
Display Name (cont.)	• European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. The maximum length of text fields also varies according to the mixture of character sets (Unicode and ASCII). Maximum field length in ASCII is 80 characters. If the same name is already used by another conference, Meeting Room or Entry Queue, the RMX displays an error message requesting you to enter a different name.
Routing Name	Enter a name using ASCII text only. If no *Routing Name* is entered, the system automatically assigns a new name as follows: • If an all ASCII text is entered in *Display Name*, it is used also as the *Routing Name.* • If any combination of Unicode and ASCII text (or full Unicode text) is entered in *Display Name*, the *ID* (such as Conference ID) is used as the *Routing Name.*
Profile	Select the Profile to be used by the Entry Queue. The default Profile is selected by default. This Profile determines the Bit Rate and the video properties with which participants connect to the Entry Queue and destination conference. In Ad Hoc conferencing it is used to define the new conference properties.
ID	Enter a unique number identifying this conferencing entity for dial in. Default string length is 4 digits. If you do not manually assign the ID, the MCU assigns one after the completion of the definition. The ID String Length is defined by the flag NUMERIC_CONF_ID_LEN in the System Configuration.

Table 4-2: Entry Queue Definitions Parameters (Continued)

Option	Description
Entry Queue IVR Service	The default Entry Queue IVR Service is selected. If required, select an alternate Entry Queue IVR Service, which includes the required voice prompts, to guide participants during their connection to the Entry Queue.
Ad Hoc	Select this check box to enable the Ad Hoc option for this Entry Queue.
Cascade	Set this field to **None** for all Entry Queues other than cascading. If this Entry Queue is used to connect dial-in cascaded links, select **Master** or **Slave** depending on the Master/ Slave relationship in the Cascading topology. Set this field to *Master* if: • The Entry Queue is defined on the MCU on level 1 and the dialing is done from level 2 to level 1. • The Entry Queue is defined on the MCU on level 2 and the dialing is done from level 3 to level 2. Set this field to *Slave* if the Entry Queue is defined on the MCU on level 2 (Slave) and the dialing is done from MCU level 1 to level 2.
Enable ISDN/ PSTN Access	Select this check box to allocate dial-in numbers for ISDN/PSTN connections. To define the first dial-in number using the default ISDN/PSTN Network Service, leave the default selection. When the Entry Queue is saved on the MCU, the dial-in number will be automatically assigned to the Entry Queue. This number is taken from the dial-in numbers range in the default ISDN/PSTN Network Service.
ISDN/PSTN Network Service	The default Network Service is automatically selected. To select a different ISDN/PSTN Network Service in the service list, select the name of the Network Service.

Table 4-2: Entry Queue Definitions Parameters (Continued)

Option	Description
Dial-in Number (1)	Leave this field blank to let the system automatically assign a number from the selected ISDN/PSTN Network Service. To manually define a dial-in number, enter a required number from the dial-in number range defined for the selected Network Service.
Dial-in Number (2)	By default, the second dial-in number is not defined. To define a second-dial-in number, enter a required number from the dial-in number range defined for the selected Network Service.

4 Click **OK**.

The new *Entry Queue* is added to the *Entry Queues* list.

Listing Entry Queues

To view the list of Entry Queues:

- In the *RMX Management - Rarely Used* pane, click **Entry Queues**.

The *Entry Queues* are listed in the *Entry Queues* pane.

You can double-click an Entry Queue to view its properties.

Modifying the EQ Properties

To modify the EQ:

- In the *Entry Queues* pane, either double-click or right-click and select **Entry Queue Properties** of the selected *Entry Queue* in the list.

  The Entry Queue Properties dialog box is displayed. All the fields may be modified except **Routing Name**.

Transit Entry Queue

A *Transit Entry Queue* is an Entry Queue to which calls with dial strings containing incomplete or incorrect conference routing information are transferred.

IP Calls are routed to the *Transit Entry Queue* when:

- A gatekeeper is not used, or where calls are made directly to the RMX's *Signaling IP Address*, with incorrect or without a Conference ID.
- When a gatekeeper is used and only the prefix of the RMX is dialed, with incorrect or without a Conference ID.
- When the dialed prefix is followed by an incorrect conference ID.

When no *Transit Entry Queue* is defined, all calls containing incomplete or incorrect conference routing information are rejected by the RMX.

In the *Transit Entry Queue*, the *Entry Queue IVR Service* prompts the participant for a destination conference ID. Once the correct information is entered, the participant is transferred to the destination conference.

Setting a Transit Entry Queue

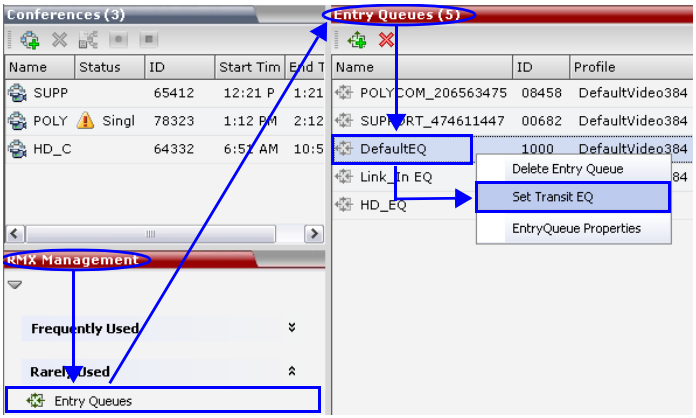
The RMX factory default settings define the *Default Entry Queue* also as the *Transit Entry Queue*. You can designate another Entry Queue as the *Transit Entry Queue*.

Only one *Transit Entry Queue* may be defined per RMX and selecting another Entry Queue as the *Transit Entry Queue* automatically cancels the previous selection.

To designate an Entry Queue as Transit Entry Queue:

1. In the *RMX Management - Rarely Used* pane, click **Entry Queues**.

2 In the *Entry Queues* list, right-click the Entry Queue entry and then click **Set Transit EQ.**



The Entry Queue selected as *Transit Entry Queue* is displayed in bold.

To cancel the Transit Entry Queue setting:

1 In the *RMX Management - Rarely Used* pane click **Entry Queues**.

2 In the *Entry Queues* list, right-click the *Transit Entry Queue* entry and then click **Cancel Transit EQ**.

Ad Hoc Conferencing

The Entry Queue can also be used for Ad Hoc conferencing. If the Ad Hoc option is enabled for the Entry Queue, when the participant enters the target conference ID the system checks whether a conference with that ID is already running on the MCU. If not, the system automatically creates a new ongoing conference with that ID. The conference parameters are based on the Profile linked to the Entry Queue. As opposed to Meeting Rooms, that are predefined conferences saved on the MCU, Ad Hoc conferences are not stored on the MCU. Once an Ad Hoc conference is started it becomes an ongoing conference, and it is monitored and controlled as any standard ongoing conference.

An external database application can be used for authentication with Ad Hoc conferences. The authentication can be done at the Entry Queue level and at the conference level. At the Entry Queue level, the MCU queries the external database server whether the participant has the right to create a new conference. At the conference level the MCU verifies whether the participant can join the conference and if the participant is the conference chairperson. The external database can populate certain conference parameters.

For more information about Ad Hoc conferencing, see *Appendix D: "Ad Hoc Conferencing and External Database Authentication"* on page **D-1**.

Gateway to Polycom® Distributed Media Application™ (DMA™) 7000

Gateway to Polycom® Distributed Media Application™ (DMA™) 7000 enables audio only PSTN, ISDN (video endpoints using only their audio channels), SIP and H.323 calls can connect to the Polycom DMA 7000 via gateway sessions running on the RMX. Each RMX conference acting as a gateway session includes one connection to the endpoint and another connection to the DMA. The DMA 7000 enables load balancing and the distribution of multipoint calls on up to 10 Polycom RMX media servers.

As part of this solution, the RMX acts as a gateway for the DMA that supports H.323 calls. The PSTN, ISDN or SIP endpoint dials the virtual Meeting Room on the DMA via a special Entry Queue on the RMX.

For more information, see Appendix H, "*Gateway to Polycom® DMA™ 7000*".

SIP Factories

A SIP Factory is a conferencing entity that enables SIP endpoints to create Ad Hoc conferences. The system is shipped with a default SIP Factory, named DefaultFactory.

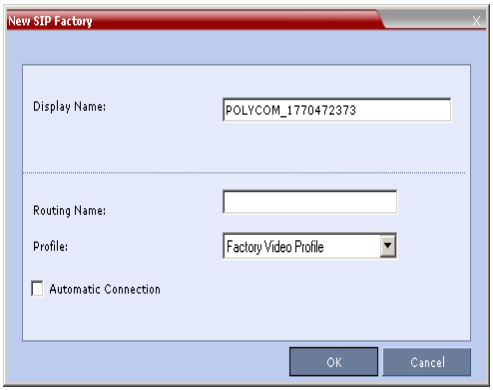
When a SIP endpoint calls the SIP Factory URI, a new conference is automatically created based on the Profile parameters, and the endpoint joins the conference.

The SIP Factory URI must be registered with the SIP server to enable routing of calls to the SIP Factory. To ensure that the SIP factory is registered, the option to register *Factories* must be selected in the Default IP Network Service.

Creating SIP Factories

To create a new SIP Factory:

1. In the *RMX Management - Rarely Used* pane, click **SIP Factories**.
2. In the *SIP Factories* list pane, click the **New SIP Factory** button. The *New Factory* dialog box opens.



3 Define the following parameters:

Table 4-3: New Factory Properties

Option	Description
Display Name	Enter the SIP Factory name that will be displayed. The Display Name is the conferencing entity name in native language character sets to be displayed in the RMX Web Client. In conferences, Meeting Rooms, Entry Queues and SIP factories the system automatically generates an ASCII name for the *Display Name* field that can be modified using Unicode encoding. • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. The maximum length of text fields also varies according to the mixture of character sets (Unicode and ASCII). Maximum field length in ASCII is 80 characters. If the same name is already used by another conference, Meeting Room or Entry Queue, the RMX displays an error message requesting you to enter a different name.
Routing Name	The *Routing Name* is defined by the user, however if no *Routing Name* is entered, the system will automatically assign a new name when the Profile is saved as follows: • If an all ASCII text is entered in *Display Name*, it is used also as the *Routing Name.* • If any combination of Unicode and ASCII text (or full Unicode text) is entered in *Display Name*, the *ID* (such as Conference ID) is used as the *Routing Name.*

Table 4-3: New Factory Properties (Continued)

Option	Description
Profile	The default Profile is selected by default. If required, select the conference Profile from the list of Profiles defined in the MCU. A new conference is created using the parameters defined in the Profile.
Automatic Connection	Select this check box to immediately accept the conference creator endpoint to the conference. If the check box is cleared, the endpoint is redirected to the conference and then connected.

4 Click **OK**.
The new SIP Factory is added to the list.

5

Address Book

The Address Book is your database and information storage for the people and businesses you communicate with. The Address Book stores, among many other fields, IP addresses, phone numbers and network communication protocols used by the participant's endpoint. By utilizing the Address Book users are able to quickly and efficiently assign or designate participants to conferences. Groups defined in the Address Book help facilitate the creation of conferences. Rather than adding each participant individually to a conference, groups enable multiple participants to be added to a conference.

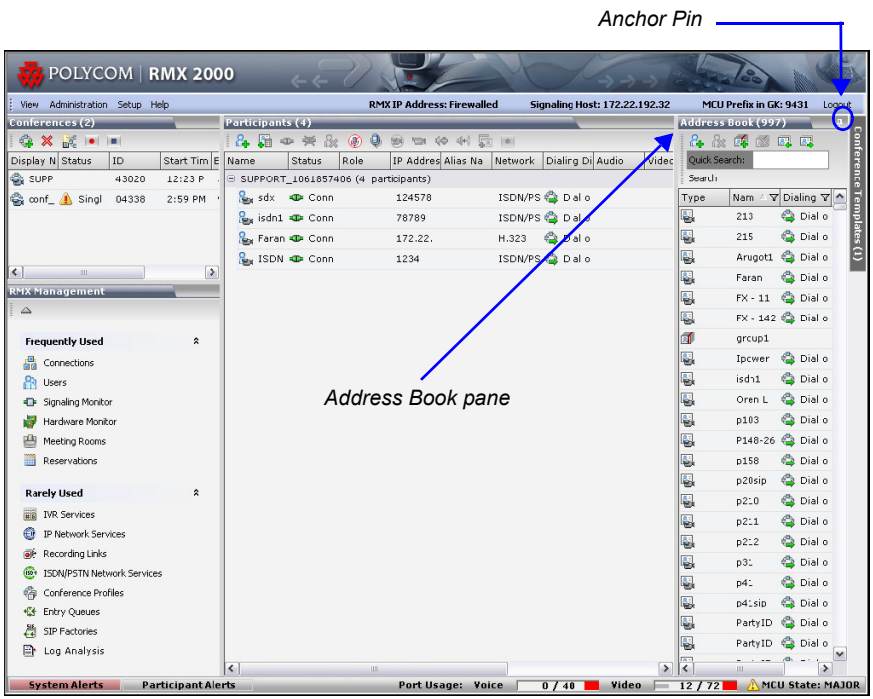
The maximum number of entries in the RMX internal Address Book is 1000. When using the Polycom CMA Global Address Book, all entries are listed.

Importing and exporting of Address Books enables organizations to seamlessly distribute up-to-date Address Books to multiple RMX units. It is not possible to distribute Address Books to external databases running on applications such as *Polycom's ReadiManager (SE200)* or *Polycom CMA*. External databases can run in conjunction with RMX units, but must be managed from the external application, e.g. new participants cannot be added to the external database from the RMX Web Client. To enable the RMX to run with an external database such as Polycom CMA, the appropriate system configuration flags must be set. For more information, see "*System Configuration*" on page **16-10**.

Integration with Polycom CMA Global Address Book is supported. For details, see "*Integrating the Polycom CMA™ Address Book with the RMX*" on page **5-22**. Integration with the *SE200* GAB (Global Address Book) is not supported.

Viewing the Address Book

You can view the participants currently defined in the Address Book. The first time the *RMX Web Client* is accessed, the *Address Book* pane is displayed.

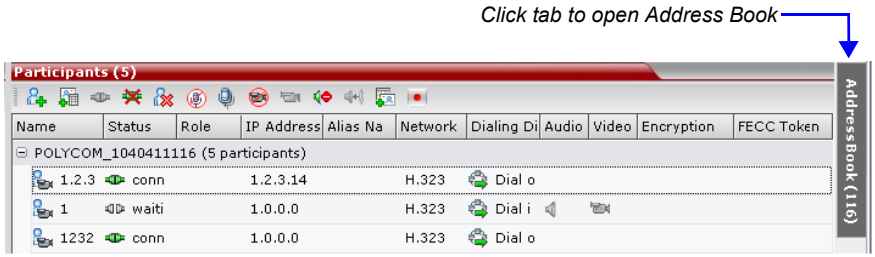


Displaying and Hiding the Address Book

The Address Book can be hidden it by clicking the anchor pin ( ) button in the pane header.

The *Address Book* pane closes and a tab appears at the right edge of the screen.

Click the tab to re-open the *Address Book*.



The following information is displayed for each participant. The fields displayed vary accordingly, when viewing the full display or the docked Address Book pane.

Table 5-1 *Docked Address Book List Columns*

Field/Option	Description
Type	Indicates whether the participant is a video ( ) or audio ( ).
Name	Displays the name of the participant.
IP Address/Phone	Indicates the IP address and phone number of the participant's endpoint. For SIP participants, the IP address is displayed only if one was defined for the participant.
Dialing Direction	*Dial-in* – The participant dials in to the conference. *Dial-out* – The RMX dials out to the participant.

Adding a Participant to the Address Book

Adding participants to the Address Book can be performed by the following methods:

- Directly in the Address Book.
- Moving or saving a participant from an ongoing conference to the Address Book.

Only defined **dial-out** ISDN/PSTN participants can be added to the Address Book or ongoing conferences. ISDN/PSTN participants are added to the Address Book in the same manner that H.323 and SIP participants are added.

When adding dial-out participants to the ongoing conference, the system automatically dials out to the participants using the Network Service (ISDN/PSTN or IP) defined for the connection in the participant properties.

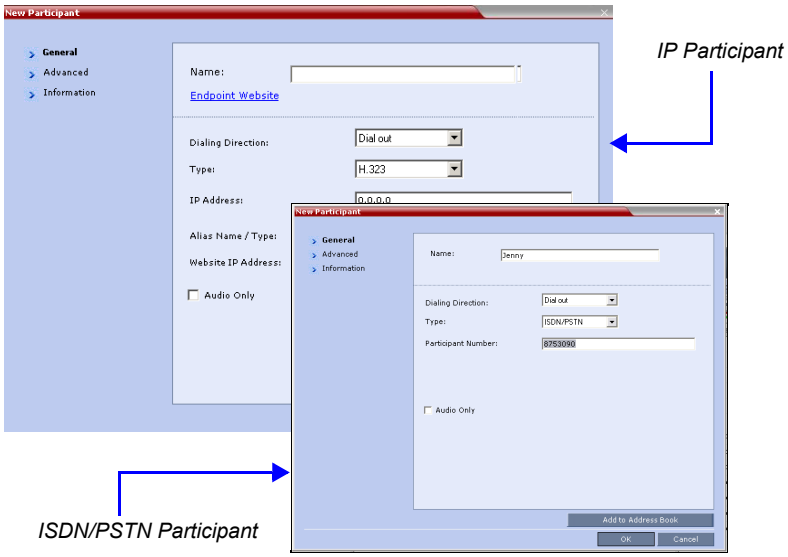
Adding a new participant to the Address Book Directly

New participants can be added directly in the Address Book as needed.

To add a new participant to the Address Book:

1 In the *Address Book* pane, click the **New Participant** button( ).

The *New Participant - General* dialog box opens.



2 Define the following fields:

Table 5-2 New Participant – General Properties

Field	Description
Name	Enter the name of the participant or the endpoint as it will be displayed in the RMX Web Client. The *Name* field can be modified using Unicode encoding. • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum.• Maximum field length in ASCII is 80 characters. The maximum length of text fields varies according to the mixture of character sets used (Unicode and ASCII). This name can also become the endpoint name that is displayed in the video layout. For more details about endpoint (site) names, see the *RMX Getting Started Guide*, *""Text Indication in the Video Layout"* on page 3-36". **Note:** This field is displayed in all tabs.
Endpoint Website (IP only)	Click the Endpoint Website hyperlink to connect to the internal website of the participant's endpoint. It enables you to perform administrative, configuration and troubleshooting activities on the endpoint. The connection is available only if the IP address of the endpoint's internal site is defined in the *Website IP Address* field.

Table 5-2 New Participant – General Properties (Continued)

Field	Description
Dialing Direction	Select the dialing direction: • ***Dial-in*** – The participant dials in to the conference. • ***Dial-out*** – The MCU dials out to the participant. **Notes:** • This field applies to IP participants only. • Dial-out is forced when defining an ISDN/PSTN participant.
Type	The network communication protocol used by the endpoint to connect to the conference: *H.323*, *SIP* or *ISDN/PSTN*. The fields in the dialog box change according to the selected network type.
IP Address **(H.323 and SIP Only)**	Enter the IP address of the participant's endpoint. • For H.323 participant define either the endpoint IP address or alias. • For SIP participant define either the endpoint IP address or the SIP address. **Note:** This field is hidden when the ISDN/PSTN protocol is selected.
Phone Number **(ISDN/PSTN Only)**	Enter the phone number of the ISDN/PSTN participant. **Note:** This field is only displayed when the ISDN/PSTN protocol is selected.

Table 5-2 New Participant – General Properties (Continued)

Field	Description
Alias Name/Type **(H.323 Only)**	If you are using the endpoint's alias and not the IP address, first select the type of alias and then enter the endpoint's alias: • H.323 ID (alphanumeric ID) • E.164 (digits 0-9, * and #) • Email ID (email address format, e.g. abc@example.com) • Participant Number (digits 0-9, * and # **Note:** • Although all types are supported, the type of alias is dependent on the gatekeeper's capabilities. The most commonly supported alias types are H.323 ID and E.164. • This field is used to enter the Entry Queue ID, target Conference ID and Conference Password when defining a cascaded link. • This field is removed from the dialog box when the ISDN/PSTN protocol is selected.

Table 5-2 New Participant – General Properties (Continued)

Field	Description
SIP Address/ Type **(SIP Only)**	Select the format in which the SIP address is written: • ***SIP URI*** - Uses the format of an E-mail address, typically containing a user name and a host name: *sip:[user]@[host]*. For example, sip:dan@polycom.com. • ***TEL URI*** - Used when the endpoint does not specify the domain that should interpret a telephone number that has been input by the user. Rather, each domain through which the request passes would be given that opportunity. As an example, a user in an airport might log in and send requests through an outbound proxy in the airport. If the users enters “411” (this is the phone number for local directory assistance in the United States), this number needs to be interpreted and processed by the outbound proxy in the airport, and not by the user's home domain. In this case, tel: 411 is the correct choice. **Note:** This field is removed from the dialog box when the ISDN/PSTN protocol is selected.
Endpoint Website IP Address (IP only)	Enter the IP address of the endpoint’s internal site to enable connection to it for management and configuration purposes. This field is automatically completed the first time that the endpoint connects to the RMX. If the field is blank it can be manually completed by the system administrator. The field can be modified while the endpoint is connected
Audio Only	Select this check box to define the participant as a voice participant, with no video capabilities.

3 Usually, additional definitions are not required and you can use the system defaults for the remaining parameters. In such a case, click **OK**.

To modify the default settings for advanced parameters, click the **Advanced** tab.

4 Define the following *Advanced* parameters:

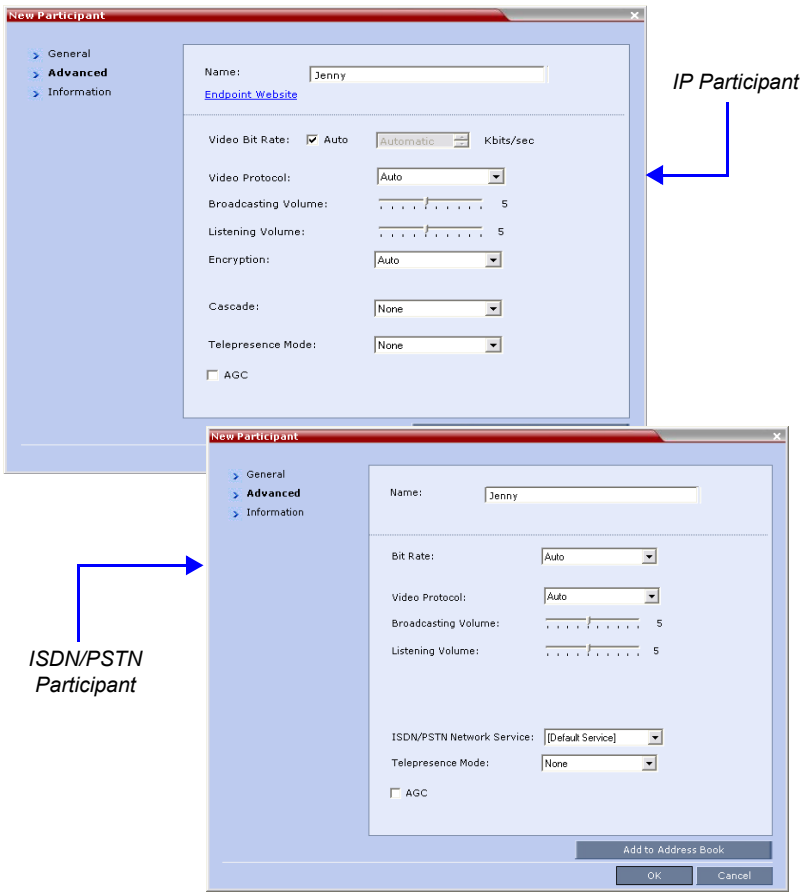


Table 5-3 *New Participant – Advanced Properties*

Field	Description
Video Bit Rate / Auto **(IP Only)**	The *Auto* check box is automatically selected to use the Line Rate defined for the conference. **Note:** This check box cannot be cleared when defining a new participant during an ongoing conference. To specify the video rate for the endpoint, clear this check box and then select the required video rate.

Table 5-3 New Participant – Advanced Properties (Continued)

Field	Description
Video Protocol	Select the video compression standard that will be forced by the MCU on the endpoint when connecting to the conference: *H.261*, *H.263* or *H.264*. Select **Auto** to let the MCU select the video protocol according to the endpoint's capabilities.
Broadcasting Volume + *Listening Volume*	To adjust the volume the participant broadcasts to the conference or the volume the participant hears the conference, move the slider; each unit represents an increase or decrease of 3 dB (decibel). The volume scale is from 1 to 10, where 1 is the weakest and 10 is the strongest. The default connection value is 5.
Encryption **(IP Only)**	Select whether the endpoint uses encryption for its connection to the conference. **Auto** (default setting) indicates that the endpoint will connect according to the conference encryption setting. Encryption is not supported in ISDN/PSTN calls. ISDN/PSTN participants can connect to encrypted conferences only if the system flag is set to allow non-encrypted participants to connect to encrypted conferences.
AGC	AGC (Auto Gain Control) mechanism regulates noise and audio volume by keeping the received audio signals of all participants balanced. Select this check box to enable the AGC mechanism for participants with weaker audio signals. **Note:** Enabling AGC may result in amplification of background noise.

Table 5-3 New Participant – Advanced Properties (Continued)

Field	Description
Cascaded Link ***(IP Only)***	If this participant is used as a link between conferences select: • **Slave**, if the participant is defined in a conference running on a Slave MCU. • **Master**, if the participant is defined in a conference running on the Master MCU. It enables the connection of one conference directly to another conference using an H.323 connection only. The conferences can run on the same MCU or different MCU's. For more information, see "*Enabling Cascading"* on page **2-51**.
ISDN/PSTN Network Service	Enables users to select the ISDN/PSTN network service.
Telepresence Mode	Setting the participant/endpoint Telepresence Mode configures the RMX to receive the video format of the RPX or TPX room endpoints. If you are defining an endpoint that is part of a telepresence room, select the room type as follows: • **RPX** - for room endpoints that transmit 4:3 video format. • **TPX** - for room endpoints that transmit 16:9 video format. • **None (default)** - to indicate a standard endpoint that is not part of a telepresence room configuration.

5 To add general information about the participant, i.e. email, company name, etc..., click the **Information** tab and type the necessary details in the **Info 1-4** fields. Text in the *info* fields can be added in Unicode format (length: 31 characters).

6 Click **OK**.

The new participant is added to the address book.

Adding a Participant from an Ongoing Conference to the Address Book

You can add a participant to the Address Book directly from an ongoing conference.

To add a participant from the conference to the Address Book:

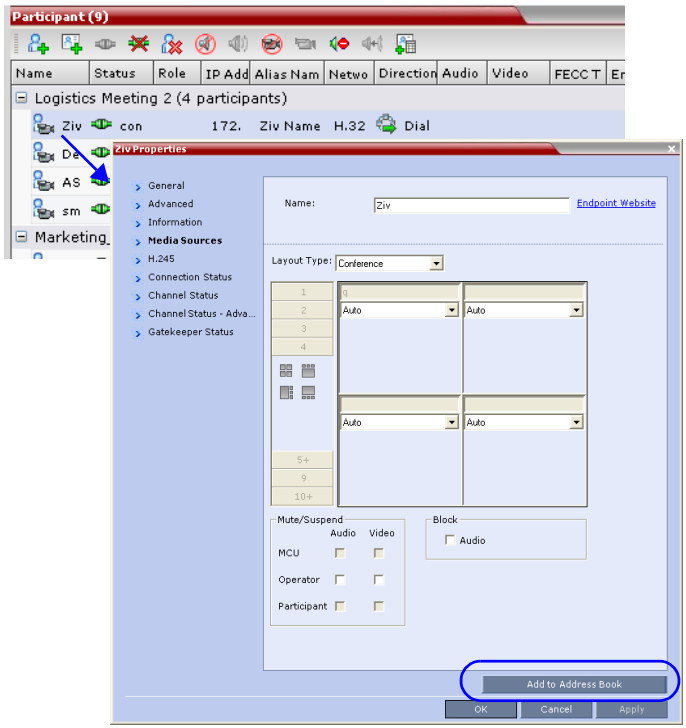
1 During an ongoing conference, select the participant in the *Participant* pane and either click the **Add Participant to Address Book** button ( ) or right-click and select **Add Participant to Address Book**.

The participant is added to the Address Book.

Alternatively, you could:

a Double-click the participant's icon or right-click the participant icon and click **Participant Properties**.

The *Participant Properties* window opens.



b Click the **Add to Address book** button.

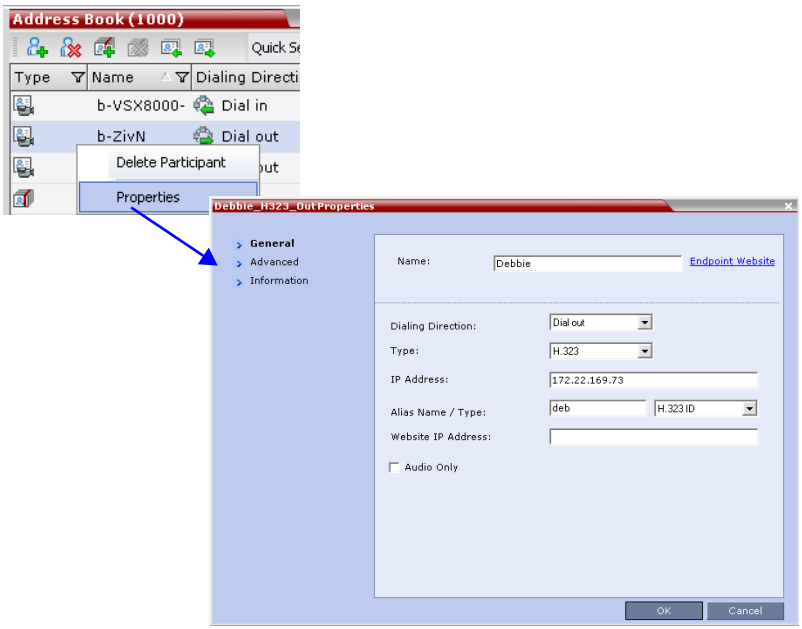
Modifying Participants in the Address Book

When required, you can modify the participant's properties.

To modify participant properties in the Address Book:

1 In the *Address Book* pane, double-click the participant's icon or right-click the participant's name and click **Participant Properties**.

The *Participant's Properties* window appears.



2 Modify the necessary properties in the window, e.g. dialing direction, communication protocol type, etc... You can modify any property in any of the three tabs: *General*, *Advanced* and *Info*.

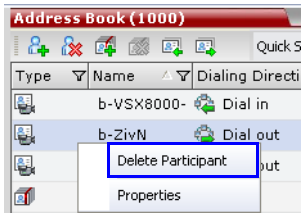
3 Click **OK**.

The changes to the participant's properties are updated.

Deleting Participants from the Address Book

To delete participants from the Address Book:

1 In the *Address Book* pane select the participant to delete. Click the **Delete Participant** ( ) button or right-click and then click the **Delete Participant** option.



2 Click **Yes** in the dialog box that appears to confirm the deletion.

After confirmation, the selected participant is deleted from the Address Book.

Searching the Address Book

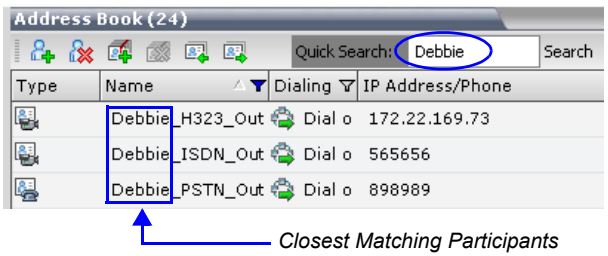
To search for participants in the Address Book:

1 In the *Address Book* toolbar, click in the *Quick Search* field.

The field clears and a cursor appears indicating that the field is active.

2 Type all or part of the participant's *Name* and click **Search**.

The closest matching participant entries are displayed and the *Active Filter* indicator turns on.



Filtering the Address Book

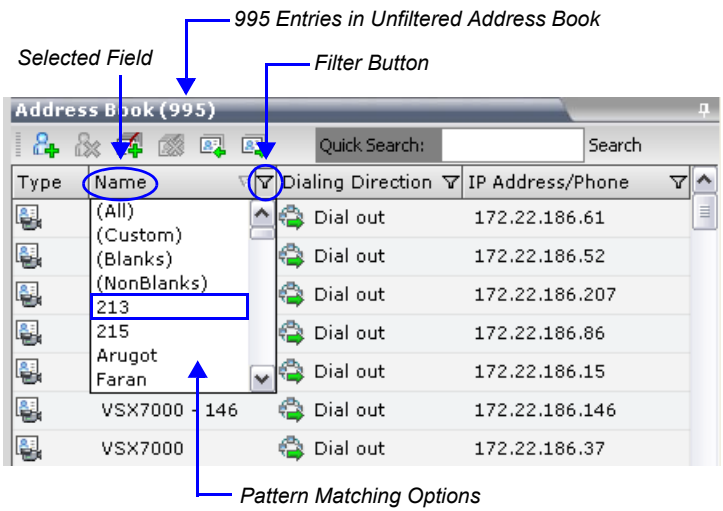
Filtering applies pattern matching criteria to the information in the *Address Book* entries, enabling you to select and work with a subset of *Address Book* entries.

Filtering can be applied to one or multiple *Address Book* fields at a time.

To filter an address book field:

1 In the *Address Book* field that you want to filter, click the filter (▽) button.

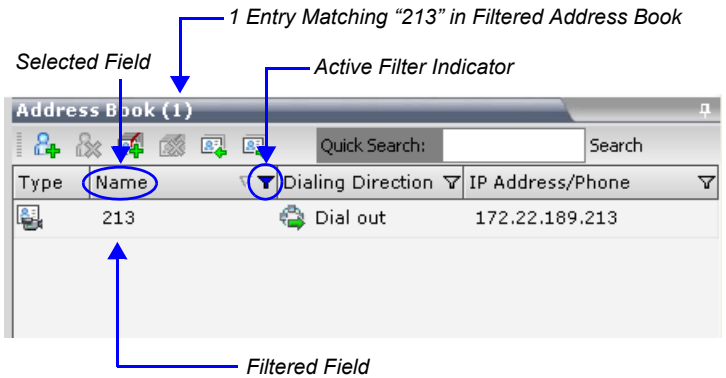
A drop-down menu is displayed containing all the matching patterns that can be applied to the selected field.



2 Click the matching pattern to be applied as the filter.

The filtered list is displayed with an active filter (blue) indicator (▼) displayed in the selected field heading.

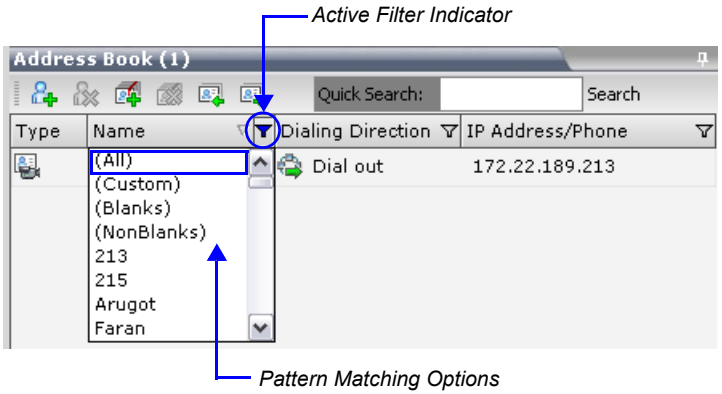
Example: If the user selects **213** as the matching pattern, the filtered *Address Book* is displayed as follows:



To clear the filter and display all entries:

1 In the filtered *Address Book* field heading, click the *Active Filter* indicator.

The pattern matching options menu is displayed.



2 Click **All**.

The filter is de-activated and all *Address Book* entries are displayed.

Participant Groups

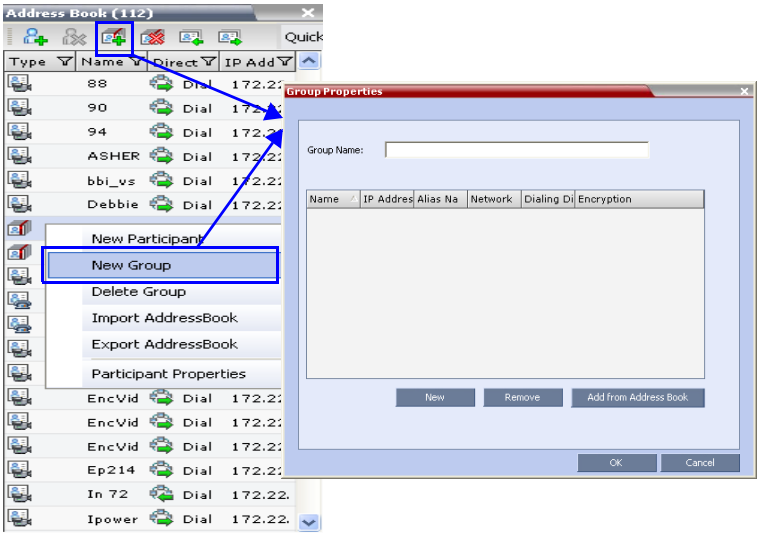
A group is a predefined collection of participants. A group provides an easy way to connect a combination of endpoints to a conference. For example, if you frequently conduct conferences with the marketing department, you can create a group called "Marketing Team" that contains the endpoints of all members of the marketing team.

Adding a New Group to the Address Book

To define a New Group:

1 In the *Address Book* pane click the **New Group** ( ) button *or* right-click an empty area in the pane and click **New Group**.

The *Group Properties* dialog box appears.



2 In the *Name* field, enter a name for the group, for example, Marketing Team.

3 Add participants to the Group by doing one of the following:

- **a** Click the **Add from Address Book** button to display the *Participants Address Book* dialog box. Select the desired endpoints to include in the Group and click **Add**. Multiple selections of participants are enabled.
- **b** Drag and drop the desired endpoints from the *Address Book* pane into the Group's dialog box.
- **c** Click the **New** button to display the *New Participant* dialog box. Define the endpoint's parameters and click **OK**.

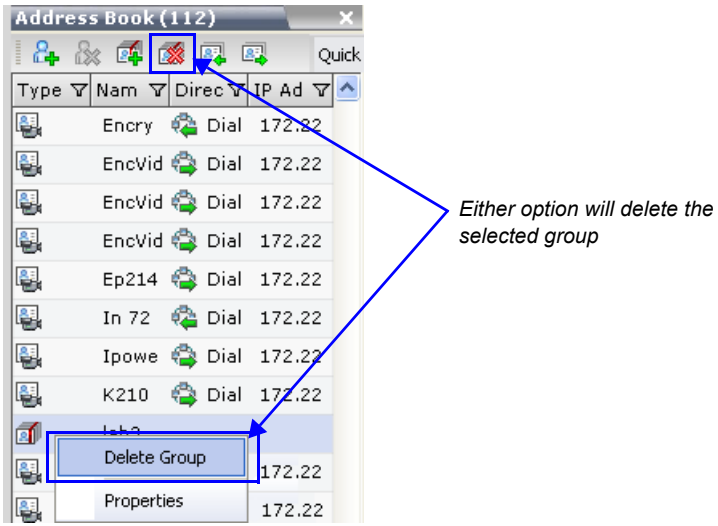
4 In the *Group* dialog box, click **OK**.

The new group is added to the *Address Book*.

Deleting a Group from the Address Book

To delete a Group:

1 In the *Address Book* pane, select the group and click the **Delete Group** button or right-click the group and click the **Delete Group**.



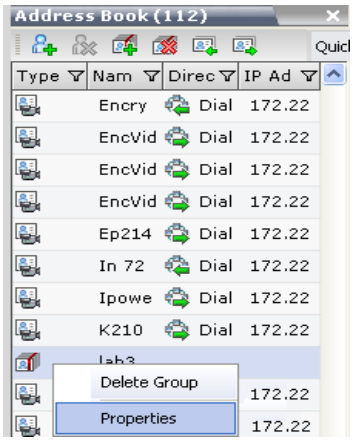
2 Click **Yes** in the confirmation dialog box.

The selected group is deleted from the *Address Book*.

Modifying a Group in the Address Book

To Modify a Group:

1 In the *Address Book* pane, double-click the Group icon ( ) or right-click the Group and then click **Properties**.



The *Group Properties* dialog box will be displayed.

2 The following operation can be performed:

- **a** **Rename Group** – Rename the Group in the name field.
- **b** **Create New Participant** – Click the **New** button to create new participants in the *Address Book* and included them in the Group.
- **c** **Add Participant** – Add one or more participants to the Group by clicking the **Add from Address Book** button and selecting the participants from the *Participants Address Book* dialog box.
- **d** **Remove Participant** – Select the one or more participants in the *Group properties* dialog box and click the **Remove** button.

Standard Windows multiple selection techniques can be for (adding/removing) participants (to/from) the Group.

3 Click **OK**.

Importing and Exporting Address Books

Address Books are proprietary Polycom data files that can only be distributed among RMX units. The Address Books are exported in XML format, which are editable offline. If no name is assigned to the exported Address Book, the default file name is:

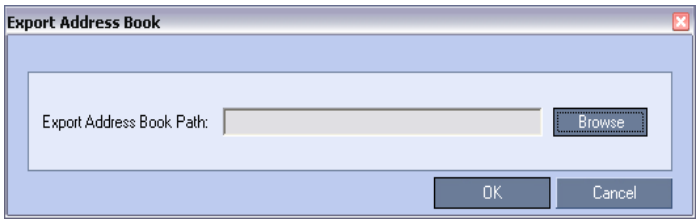
`EMA.DataObjects.OfflineTemplates.AddressbookContent_.xml`

Exporting an Address Book

To Export an Address Book:

1. In the *Address Book* pane, click the **Export Address Book** ( ) button or right-click an empty area in the pane and click **Export Address Book**.

   The *Export Address Book* dialog box appears.



2. Enter the desired path or click the **Browse** button.
3. In the **Save Address Book** dialog box, select the directory to save the file. You may also rename the file in the *File Name* field.
4. Click **Save**.

   You will return to the *Export File* dialog box.
5. Click **OK**.

   The exported Address Book is saved in the selected folder in XML format.

Importing an Address Book

To Import and Address Book:

1. In the *Address Book* pane, click the **Import Address Book** ( ) button or right-click an empty area in the pane and then click **Import Address Book**.

   The *Import Address Book* dialog box appears.

2. Enter the path from which to import the Address Book or click the **Browse** button.
3. In the *Open* dialog box navigate to the desired Address Book file (in XML format) to import.

When importing an Address Book, participants with exact names in the current Address Book will be overwritten by participants defined in the imported Address Book.

4. Click **Open**.

   You will return to the *Import File* dialog box.
5. Click **OK**.

   The *Address Book* is imported and a confirmation message is displayed at the end of the process.
6. Click **Close**.

Integrating the Polycom CMA™ Address Book with the RMX

The Polycom CMA™ application includes a Global Address Book with all registered endpoints. This address book can be used by the RMX 2000 to add participants to conferences.

CMA™ Address Book Integration Guidelines

- Only one address book can be used at any time. When the CMA address book is integrated into the RMX, it replaces the RMX internal address box.
- CMA address book is used in read-only mode in the RMX. CMA Address book entries can be added or modified from the CMA application or when the endpoints register with the CMA that acts as a gatekeeper.

The RMX acts as a proxy to all address book requests between the RMX Web Client and the CMA. **Ensure that firewall and other network settings allow the RMX access to the CMA server.**

To Integrate the Polycom CMA™ Address Book with the RMX:

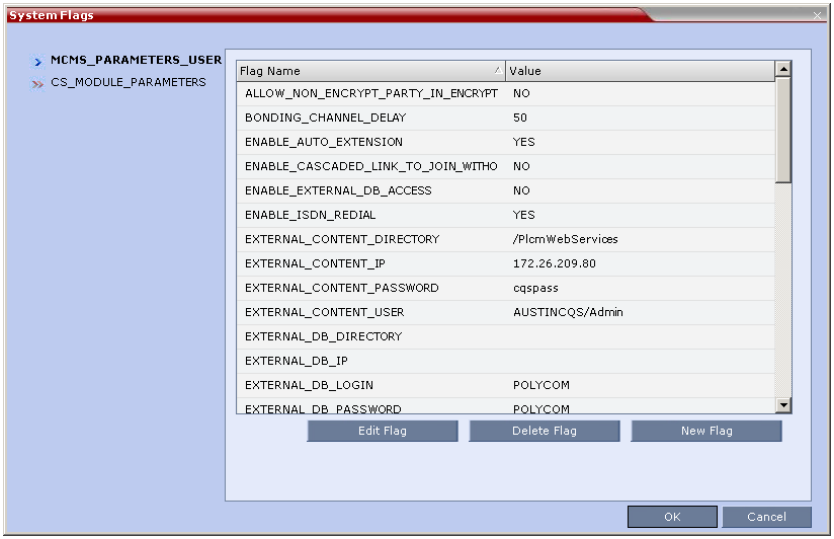
CMA Side

1. In the CMA application manually add the Polycom RMX system to the Polycom CMA system as directed in the *Polycom CMA Operations Guide*.
2. In the CMA application, add a user or use an existing user for RMX login as directed in the *Polycom CMA Operations Guide*. Write down the User Name and Password as they will be used later to define the RMX connection to the CMA Global Address Book.

RMX Side

1. On the *RMX* menu, click **Setup > System Configuration**.

The *System Flags - MCMS_PARAMETERS_USER* dialog box opens.



2 Modify the values of the flags listed below. For more details, see the RMX 2000 Administrator's Guide, "*Modifying System Flags*" on page **16-10**.

In versions 3.0 and lower, these flags have to be manually added to the MCMS_PARAMETERS_USER dialog box. In version 4.0 and higher, these flags are automatically listed in the MCMS_PARAMETERS_USER dialog box.

Table 5-4 *System Flags for CMA Address Book Integration*

Flag	Description
EXTERNAL_ CONTENT_ DIRECTORY	The Web Server folder name. Change this name if you have changed the default names used by the CMA application. Default: /PlcmWebServices
EXTERNAL_ CONTENT_IP	**The IP address of the CMA server.** This flag is also the trigger for replacing the internal RMX address book with the CMA global Address Book. When empty, the integration of the CMA address book with the RMX is disabled.

Table 5-4 *System Flags for CMA Address Book Integration (Continued)*

Flag	Description
*EXTERNAL_ CONTENT_* PASSWORD	The password associated with the user name defined for the RMX in the CMA server.
EXTERNAL_ CONTENT_ USER	The login name defined for the RMX in the CMA server defined in the format: domain name/user name.

3 Click **OK** to complete the definitions.

4 When prompted, click **Yes** to reset the MCU and implement the changes to the system configuration.

6

Reservations

The *Reservations* option enables users to schedule conferences. These conferences can be launched immediately or become ongoing, at a specified time on a specified date.

Scheduling a conference reservation requires definition of conference parameters such as the date and time at which the conference is to start, the participants and the duration of the conference.

Scheduled conferences (Reservations) can occur once or repeatedly, and the recurrence pattern can vary.

Guidelines

System

- By default, the *Scheduler* is enabled by a *System Flag*. The flag prevents potential scheduling conflicts from occurring as a result of system calls from external scheduling applications such as *ReadiManager®*, *SE200 CMA™ 4000/5000* and others via the API.

  If an external scheduling application is used, the flag INTERNAL_SCHEDULER must be manually added to the System Configuration and its value must be set to NO.
  For more information see "*Modifying System Flags*" on page **16-10**.

Resources

- The maximum number of participants per reservation is determined by the availability of system resources:
  - MPM Configuration Mode: 80 participants.
  - MPM+ Configuration Mode: 200 participants (120 voice +80 CIF video).

- System resources are calculated according to the RMX's license. For more information see "*Video/Voice Port Configuration*" on page **16-39**.
- System resource availability is partially checked when reservations are created:
  - If a conference duration extension request is received from an ongoing conference, the request is rejected if it would cause a resource conflict.
  - If several reservations are scheduled to be activated at the same time and there are not enough resources for all participants to be connected:
    - The conferences are activated.
    - Participants are connected to all the ongoing conferences until all system resources are used up.
- If sufficient resources are not available in the system and a scheduled *Reservation* cannot be activated, the *Reservation* is deleted from the schedule.
- Resources for *Reservations* are calculated using the *Reserve Resources for Audio/Video Participants* fields of the *New Reservation* dialog box. For more information see "*New Reservation – Reserved Resources*" on page **6-12**.
- Resources are reserved for participants at the highest video resolution supported by the *Line Rate* specified in the conference *Profile* and up to the maximum system video resolution specified by the MAX_CP_RESOLUTION system flag.

  If the RMX is in *MPM+ Mode* and *Fixed Capacity Mode* is selected, the number of resources allocated to this type of video participant (CIF, SD, HD) is also checked. If resource deficiencies are found an error message is displayed.
- When a new *Reservation* is created in the *Reservations*, the effect of the new *Reservation* (including its recurrences) on available resources is checked. If resource deficiencies are found an error message is displayed.

  Defined dial-in or dial-out participants, Meeting Rooms, Entry Queues and new connections to Ongoing conferences are not included in the resources calculation.

Reservations

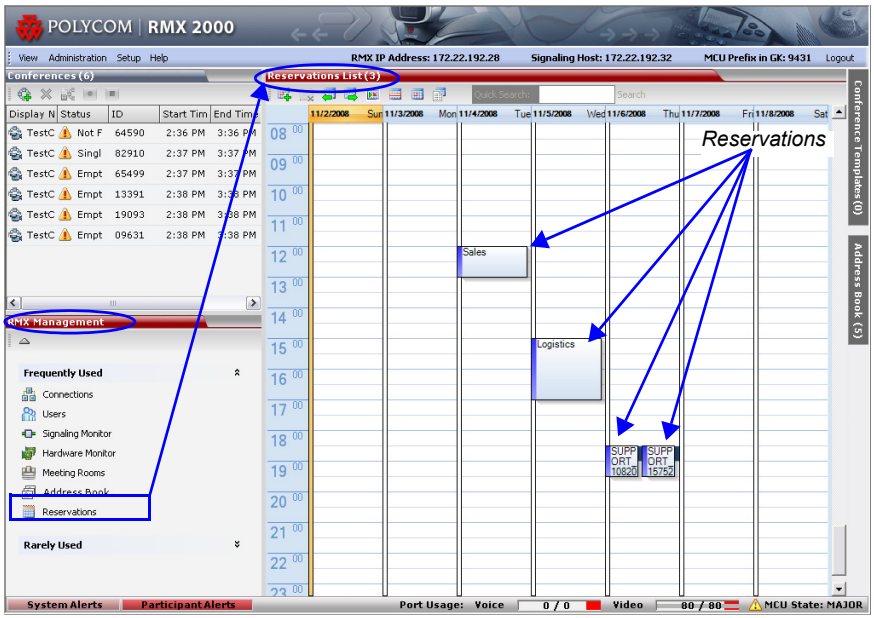
- A *Reservation* that has been activated and becomes an ongoing conference is deleted from the *Reservations* list.
- The maximum number of reservations is 2000.
- The maximum number of concurrent reservations is 80. Reservations with durations that overlap (for any amount of time) are considered to be concurrent.
- The maximum number of participants per reservation is determined by the availability of system resources:
  - MPM Configuration Mode: 80 participants
  - MPM+ Configuration Mode: 200 (120 voice +80 CIF video) participants
- System resource availability is partially checked when reservations are created:
  - If a conference duration extension request is received from an ongoing conference, the request is rejected if it would cause a resource conflict.
  - If several reservations are scheduled to be activated at the same time and there are not enough resources for all participants to be connected:
    - The conferences are activated.
    - Participants are connected to all the ongoing conferences until all system resources are used up.
- A scheduled *Reservation* cannot be activated and is deleted from the schedule if an Ongoing conference has the same *Numeric ID*.
  - Sufficient resources are not available in the system.
- If a problem prevents a *Reservation* from being activated at its schedule time, the *Reservation* will not be activated at all. This applies even if the problem is resolved during the *Reservation's* scheduled time slot.
- A Profile that is assigned to a Reservation cannot be deleted.
- Reservations are backed up and restored during **Setup > Software Management >Backup /Restore Configuration** operations. For more information see "*Software Management*" on page **16-71**.
- All existing reservations are erased by the *Standard Restore* option of the **Administration > Tools > Restore Factory Defaults** procedure.

- *Reservations* can also be scheduled from *Conference Templates*. For more information see "*Scheduling a Reservation From a Conference Template*" on page 8-16.

Using the Reservation Calendar

To open the Reservation Calendar:

- In the *RMX Management* pane, click the *Reservations* button ( ).



Toolbar Buttons

The toolbar buttons functions are described in Table 6-1.

Table 6-1 Reservations – Toolbar

Button	Description
New Reservation	Create a new reservation. The date and time of the new reservation is set according to the highlighted blocks on the *Reservations*.

Table 6-1 *Reservations – Toolbar (Continued)*

Button	Description
Delete Reservation	Click to delete the selected reservation.
Back	Click to show the previous day or week, depending on whether *Show Day* or *Show Week* is the selected.
Next	Click to show the next day or week, depending on whether *Show Day* or *Show Week* is the selected.
Today	Click to show the current date in the Reservation Calendar in either *Show Day* or *Show Week* view.
Show Week	Change the calendar view to weekly display, showing a calendar week: Sunday through Saturday
Show Day	Click this button to show the day containing the selected time slot.
Reservations List	Click to change to List View and display a list of all reservations.
Quick Search:	Used to search for reservations by *Display Name*. (Available in *Reservations List view* only).

Reservations Views

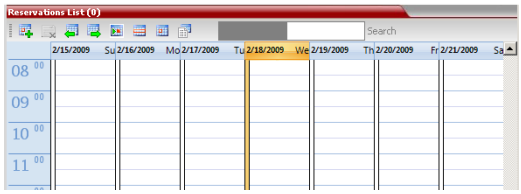
The *Reservations* has the following views available:

- Week
- Day
- Today
- List

In all views the *Main Window List Pane* header displays the total number of reservations in the system.

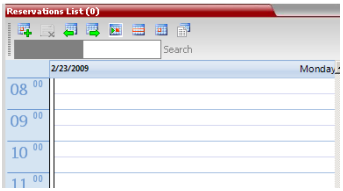
Week View

By default the *Reservations* is displayed in *Week* view with the current date highlighted in orange.



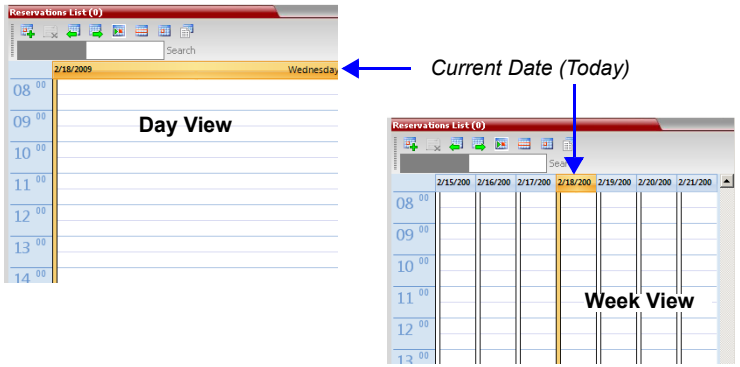
Day View

A single day is displayed.



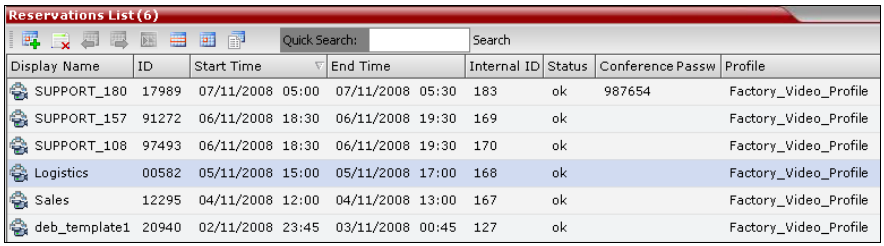
Today View

The current date (*Today*), highlighted in orange, can be viewed in both *Week View* and *Day View*.



List View

List View does not have a calendar based format.



Display Name	ID	Start Time	End Time	Internal ID	Status	Conference Passw	Profile
SUPPORT_180	17989	07/11/2008 05:00	07/11/2008 05:30	183	ok	987654	Factory_Video_Profile
SUPPORT_157	91272	06/11/2008 18:30	06/11/2008 19:30	169	ok		Factory_Video_Profile
SUPPORT_108	97493	06/11/2008 18:30	06/11/2008 19:30	170	ok		Factory_Video_Profile
Logistics	00582	05/11/2008 15:00	05/11/2008 17:00	168	ok		Factory_Video_Profile
Sales	12295	04/11/2008 12:00	04/11/2008 13:00	167	ok		Factory_Video_Profile
deb_template1	20940	02/11/2008 23:45	03/11/2008 00:45	127	ok		Factory_Video_Profile

All *Reservations* are listed by:

- *Display Name*
- *ID*
- *Internal ID*
- *Start Time*
- *End Time*
- *Status*
- *Conference Password*
- *Profile*

The *Reservations* can be sorted, searched and browsed by any of the listed fields.

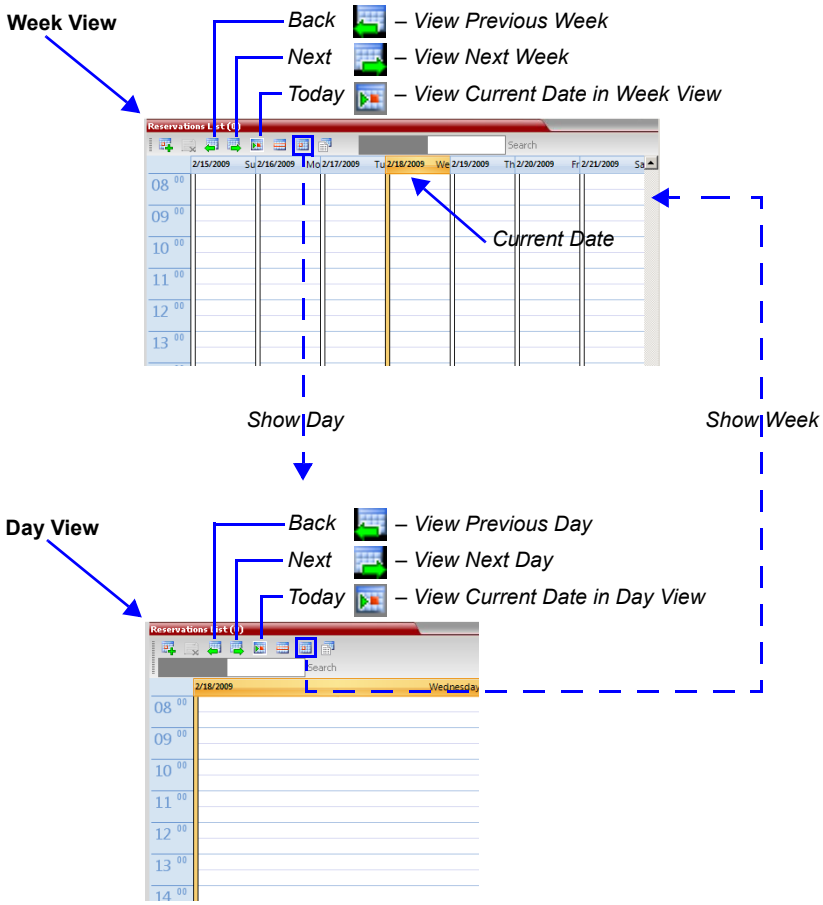
Changing the Calendar View

To change between Week and Day views:

- In Week View: In the *Reservations* toolbar, click **Show Day** ( ) to change to *Day View*.

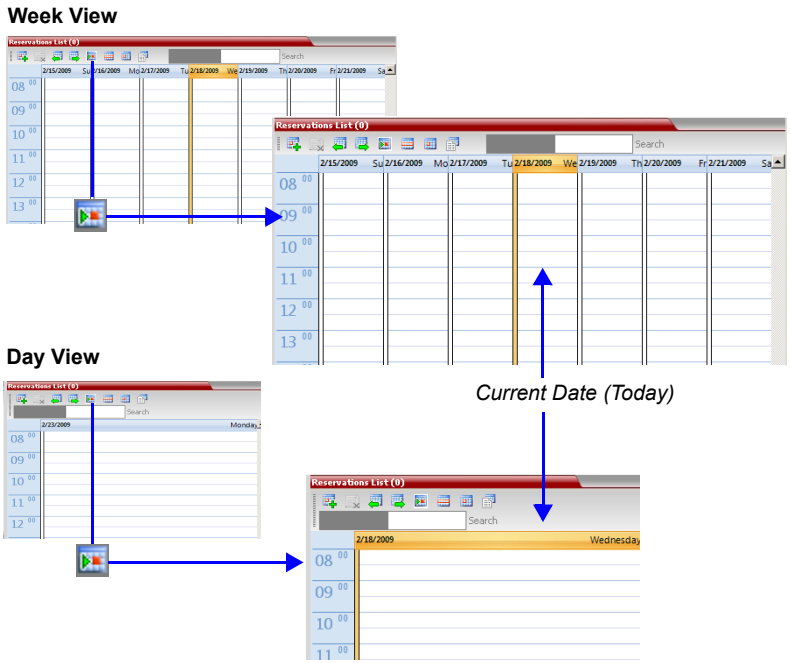
  or

  In Day View: In the *Reservations* toolbar, click **Show Week** ( ) to change to *Week View*.


Week View
Back – View Previous Week
Next – View Next Week
Today – View Current Date in Week View
Current Date
Show Day
Show Week
Day View
Back – View Previous Day
Next – View Next Day
Today – View Current Date in Day View

To view Today (the current date):

- In *Week View* or *Day View*, in the *Reservations* toolbar, click the **Today** ( ) button to have the current date displayed within the selected view.



To change to List View:

1 In the *Reservations* toolbar, click, the **Reservations List** ( ) button.

The *Reservations List* is displayed.

Display Name	ID	Start Time	End Time	Internal ID	Status	Conference Passw	Profile
SUPPORT_180	17989	07/11/2008 05:00	07/11/2008 05:30	183	ok	987654	Factory_Video_Profile
SUPPORT_157	91272	06/11/2008 18:30	06/11/2008 19:30	169	ok		Factory_Video_Profile
SUPPORT_108	97493	06/11/2008 18:30	06/11/2008 19:30	170	ok		Factory_Video_Profile
Logistics	00582	05/11/2008 15:00	05/11/2008 17:00	168	ok		Factory_Video_Profile
Sales	12295	04/11/2008 12:00	04/11/2008 13:00	167	ok		Factory_Video_Profile
deb_template1	20940	02/11/2008 23:45	03/11/2008 00:45	127	ok		Factory_Video_Profile

2 **Optional.** Sort the data by any field (column heading) by clicking on the column heading.

A ▽ or △ symbol appears in the column heading indicating that the list is sorted by this field, as well as the sort order.

3 **Optional.** Click on the column heading to toggle the column's sort order.

To return to Calendar View:

- In the *Reservations* toolbar, click any of the buttons (**Show Week/Show Day/Today**) to return to the required *Reservations* view.

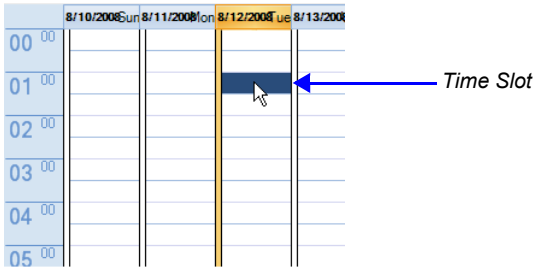
Scheduling Conferences Using the Reservation Calendar

Creating a New Reservation

There are three methods of creating a new reservation:

Each method requires the selection of a starting time slot in the *Reservations*. The default time slot is the current half-hour period of local time.

In all views, if the **New Reservation** ( ) button is clicked without selecting a starting time slot or if a time slot is selected that is in the past, the *Reservation* becomes an Ongoing conference immediately and is not added to the *Reservations* calendar.



After selecting a starting time slot in the *Reservations* you can create a reservation with a default duration derived from the creation method used or by interactively defining the duration of the reservation.

Method I – To create a reservation with default duration of 1 hour:

- In the *Reservations* toolbar, click the **New Reservation** ( ) button to create a reservation of 1 hour duration.

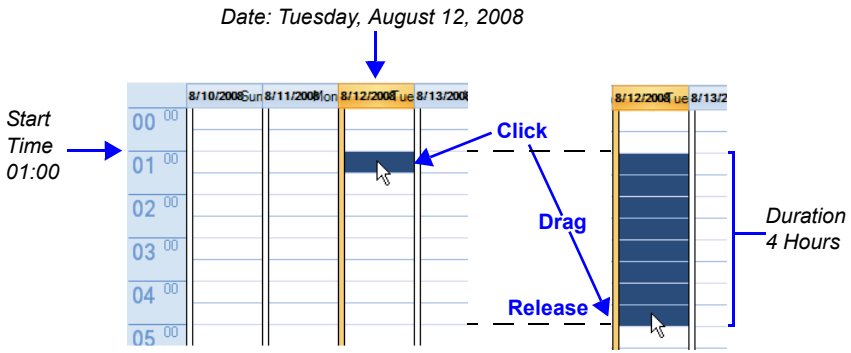
Method II – To create a reservation with default duration of ½ hour:

- Right-click and select **New Reservation** to create a reservation of ½ hour default duration.

Method III – To interactively define the duration:

1. In the calendar, click & drag to expand the time slot to select the required *Date*, *Start Time and Duration* for the reservation.
2. In the *Reservations* toolbar, click the **New Reservation** ( ) button or right-click and select **New Reservation.**

   Example: The following click & drag sequence would select a reservation for *Tuesday, August 12, 2008*, starting at *01:00* with a duration of *4* hours.

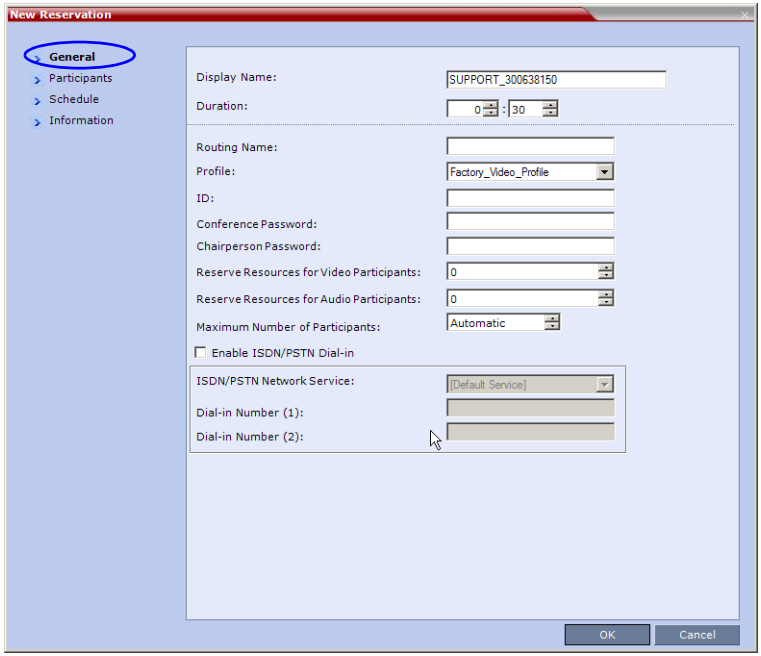


   The duration of reservations created by any of the above methods can be modified in the *Scheduler* tab of the *New Reservation* dialog box.

To create a new reservation:

1. Open the *Reservations*.
2. Select a starting time slot.
3. Create the reservation using one of the three methods described above.

The *New Reservation – General* tab dialog box opens.



All the fields are the same as for the *New Conference – General* tab, described in the *RMX 2000 Getting Started Guide*, "*General Tab*" on page **3-17**.

Table 6-2 *New Reservation – Reserved Resources*

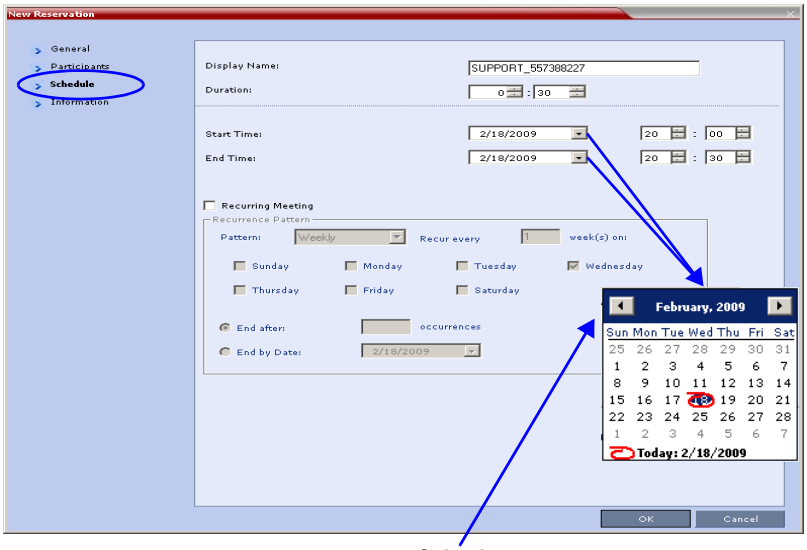
Field	Description
Reserve Resources for Video Participants	Enter the number of video participants for which the system must reserve resources. Default: 0 participants. Maximum: • MPM Mode: 80 participants. • MPM+ Mode: 80 participants.

Table 6-2 New Reservation – Reserved Resources (Continued)

Field	Description
Reserve Resources for Audio Participants	Enter the number of audio participants for which the system must reserve resources. Default: 0 participants. Maximum: • MPM Mode: 80 participants. • MPM+ Mode: 120 participants.

When a Conference Profile is assigned to a Meeting Room or a Reservation, the Profile's parameters are not embedded in the Reservation, and are taken from the Profile when the reservation becomes an ongoing conference. Therefore, any changes to the Profile parameters between the time the Reservation or Meeting Room was created and the time that it is activated (and becomes an ongoing conference) will be applied to the conference. If the user wants to save the current parameters, a different Profile with these parameters must be assigned, or a different Profile with the new parameters must be created.

4 Click the **Schedule** tab.



5 Adjust the new reservation's schedule by modifying the fields as described in Table 6-3.

Table 6-3 *New Reservation – Schedule Tab*

Field	Description	
Start Time	Select the Start Time of the Reservation.	The Start/End Times of the Reservation are initially taken from the time slot selected in the Reservation Calendar. The Start/End Times can be adjusted by typing in the hours and minutes fields or by clicking the arrow buttons. The Start/End dates can be adjusted by typing in the date field or by clicking the arrow buttons or using the calendar.
End Time	Select the End Time of the Reservation.	End Time settings are initially calculated as Start Time + Duration. End Time settings are recalculated if Start Time settings are changed. Changes to End Time settings do not affect Start Time settings. However, the Duration of the Reservation is recalculated.
Recurring Meeting	Select this option to set up a Recurring Reservation - a series of Reservations to be repeated on a regular basis. To create a recurring reservation, you must define a time period and a recurrence pattern of how often the Reservation should occur: *Daily*, *Weekly* or *Monthly*.	

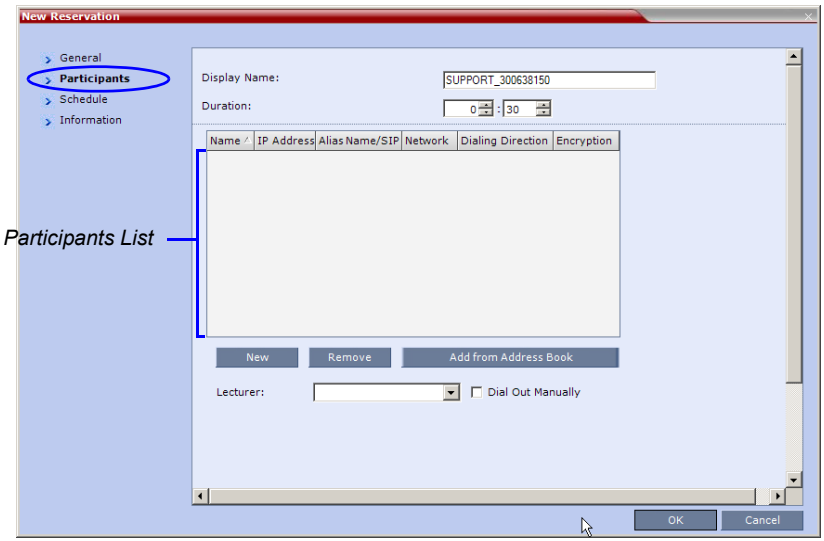
Table 6-3 *New Reservation – Schedule Tab (Continued)*

Field	Description	
Recurrence Pattern	Daily	If *Daily* is selected, the system automatically selects all the days of the week. To de-select days (for example, weekends) clear their check boxes.
	Weekly	If *Weekly* is selected, the system automatically selects the day of the week for the Reservation from the day selected in the Reservation Calendar. You can also define the recurrence interval in weeks. For example, if you want the reservation to occur every second week, enter 2 in the *Recur every _ week(s)* field. To define a twice-weekly recurring Reservation, select the check box of the additional day of the week on which the Reservation is to be scheduled and set the recurrence interval to 1.
	Monthly	If *Monthly* is selected, the system automatically selects the day of the month as selected in the Reservation Calendar. You are required to choose a recurrence pattern: • **Day (1-31) of every (1-12) month(s**) - Repeats a conference on a specified day of the month at a specified monthly interval. For example, if the first Reservation is scheduled for the 6th day of the current month and the monthly interval is set to 1, the monthly Reservation will occur on the 6th day of each of the following months. • **The (first, second,...,last) (Sun-Sat) of x month(s)** - Repeats a Reservation in a particular week, on a specified day of the week at the specified monthly interval. For example, a recurrent meeting on the third Monday every second month.

Table 6-3 *New Reservation – Schedule Tab (Continued)*

Field	Description
A series of Reservations can be set to end after a specified number of occurrences or by a specific date. Select one of the following methods of terminating the series of Reservations:	
End After	**End After: x Occurrences** - Ends a recurring series of Reservations after a specific number (x) of occurrences. Default: 1 (Leaving the field blank defaults to 1 occurrence.)
End by Date	**End By Date: mm/dd/yyyy** - Specifies a date for the last occurrence of the recurring series of Reservations. The End By Date value can be adjusted by typing in the date field or by clicking the arrow button and using the calendar utility. Default: Current date.

6 Click the **Participants** tab.



The fields are the same as for the *New Conference – Participants* tab, described in the *RMX 2000 Getting Started Guide*, "*Participants Tab*" on page **3-21**.

Participant properties are embedded in the conferencing entity and therefore, if the participant properties are modified in the *Address Book* (or *Meeting Rooms*) after the Reservation has been created they are not applied to the participant when the Reservation is activated.

7 **Optional.** Add participants from the *Participants Address Book*.

For more information see the *RMX 2000 Getting Started Guide*, "*Add from Address Book*" on page **3-23** and the *RMX 2000 Administrator's Guide*, "*Meeting Rooms*" on page **3-1**.

8 **Optional.** Add information to the reservation.

Information entered in the *Information* tab is written to the *Call Detail Record (CDR)* when the reservation is activated. Changes made to this information before it becomes an ongoing conference will be saved to the CDR.

For more information see the *RMX 2000 Getting Started Guide*, "*Information Tab*" on page **3-25**.

9 Click **OK**.
The *New Reservation* is created and is displayed in the *Reservations*.

If you create a recurring reservation all occurrences have the same ID. A recurring Reservation is assigned the same ISDN/PSTN dial-in number for all recurrences.

If a dial in number conflict occurs prior to the conference's start time, an alert appears: "ISDN dial-in number is already assigned to another conferencing entity" and the conference cannot start.

The series number (`_0000n`) of each reservation is appended to its *Display Name*.

Example:

Conference Template name: `Sales`

Display Name for single scheduled occurrence: `Sales`

If 3 recurrences of the reservation are created:

Display Name for occurrence 1: `Sales_00001`

Display Name for occurrence 2: `Sales_00002`

Display Name for occurrence 3: `Sales_00003`

Managing Reservations

Reservations can be accessed and managed via all the views of the *Reservations List*.

Guidelines

- The *Recurrence Pattern* fields in the *Schedule* tab that are used to create multiple occurrences of a *Reservation* are only displayed when the *Reservation* and its multiple occurrences are initially created.
- As with single occurrence *Reservations*, only the *Duration*, *Start Time* and *End Time* parameters of multiple occurrence reservations can be modified after the *Reservation* has been created.
- A single occurrence *Reservation* cannot be modified to become a multiple occurrence reservation.
- *Reservations* can only be modified one at a time and not as a group.
- If *Reservations* were created as a recurring series, the system gives the option to delete them individually, or all as series.

Viewing and Modifying Reservations

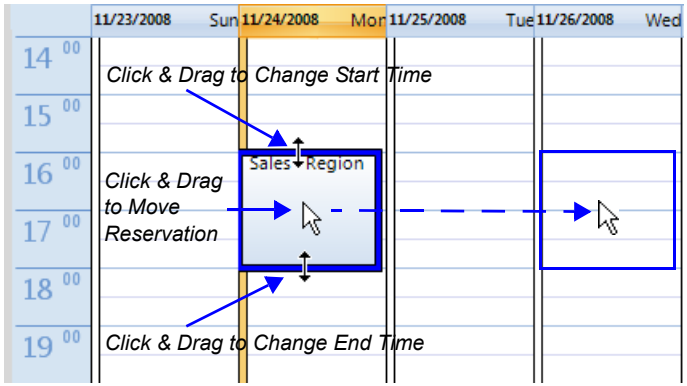
Reservations can be viewed and modified by using the *Week* and *Day* views of the *Reservations Calendar* or by using the *Reservation Properties* dialog box.

Using the Week and Day views of the Reservations Calendar

In the *Week* and *Day* views each *Reservation* is represented by a shaded square on the *Reservations*. Clicking on a *Reservation* selects the *Reservation*. A dark blue border is displayed around the edges of the *Reservation* indicating that it has been selected.

The *Start Time* of the *Reservation* is represented by the top edge of the square while the *End Time* is represented by the bottom edge.

The cursor changes to a vertical double arrow (↕) when it is moved over the top and bottom sides of the square.



To move the Reservation to another time slot:

1. Select the *Reservation*.
2. Hold the mouse button down and drag the *Reservation* to the desired time slot.
3. Release the mouse button.

To change the Reservation's Start time:

1. Select the *Reservation*.
2. Move the mouse over the top edge of the *Reservation's* square.
3. When the cursor changes to a vertical double arrow (↕) hold the mouse button down and drag the edge to the desired *Start Time*.
4. Release the mouse button.

To change the Reservation's End time:

1. Select the *Reservation*.
2. Move the mouse over the bottom edge of the *Reservation's* square.
3. When the cursor changes to a vertical double arrow (↕) hold the mouse button down and drag the edge to the desired *End Time*.
4. Release the mouse button.

To View or Modify Reservations using the Reservation Properties dialog box:

1. In the *Reservations List*, navigate to the reservation (or its recurrences) you want to view, using the **Show Day**, **Show Week**, **Today**, **Back**, **Next** or **List** buttons.
2. Double-click, or right-click and select **Reservation Properties**, to select the reservation to be viewed or modified.

   The *Reservation Properties – General* dialog box opens.
3. Select the tab(s) of the properties you want to view or modify.
4. **Optional.** Modify the *Reservation Properties*.
5. Click **OK**.
   The dialog box closes and modifications (if any) are saved.

Deleting Reservations

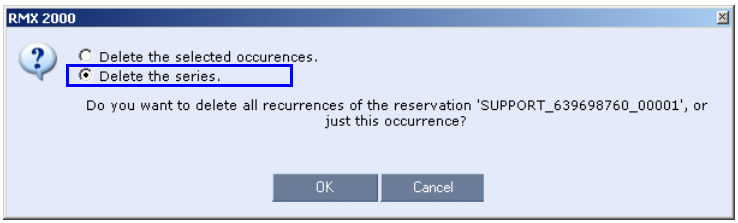
To delete a single reservation:

1. In the *Reservations List*, navigate to the reservation you want to delete, using the **Show Day**, **Show Week**, **Today**, **Back**, **Next** or **List** buttons.
2. Click to select the reservation to be deleted.
3. Click the **Delete Reservation** ( ) button.
   or
   Place the mouse pointer within the *Reservation* block, right-click and select **Delete Reservation**.
4. Click **OK** in the confirmation dialog box.
   The *Reservation* is deleted.

To delete all recurrences of a reservation:

1. In the *Reservations List*, navigate to the *Reservation* or any of its recurrences, using the **Show Day**, **Show Week**, **Today**, **Back**, **Next** or **List** buttons.
2. Click the **Delete Reservation** ( ) button.
   or
   Place the mouse pointer within the *Reservation* or any of its recurrences, right-click and select **Delete Reservation**.

A confirmation dialog box is displayed.


RMX 2000
Delete the selected occurences.
Delete the series.
Do you want to delete all recurrences of the reservation 'SUPPORT_639698760_00001', or just this occurrence?
OK
Cancel

3 Select **Delete the series.**

4 Click **OK.**

All occurrences of the *Reservation* are deleted.

Searching for Reservations using Quick Search

Quick Search is available only in *List View*. It enables you to search for *Reservations* by *Display Name*.

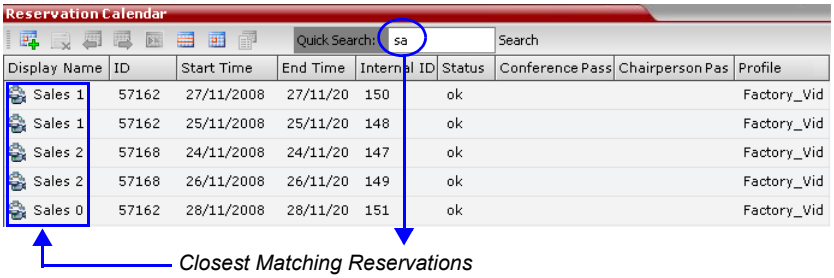
To search for reservations:

1 In the *Reservations* toolbar, click in the *Quick Search* field.

The field clears and a cursor appears indicating that the field is active.

2 Type all or part of the reservation's *Display Name* into the field and click **Search**.

The closest matching *Reservation* entries are displayed.



3 **Optional.** Double- click the *Reservation's* entry in the list to open the *Reservations Properties* dialog box to view or modify the *Reservation*.

or

Right -click the *Reservation's* entry in the list and select a menu option to view, modify or delete the *Reservation.*

To clear the search and display all reservations:

1. Clear the *Quick Search* field.
2. Click **Search.**

   All *Reservations* are displayed.

7

Operator Assistance & Participant Move

RMX users (operators) assistance to participants is available when:

- Participants have requested individual help (using *0 DTMF code) during the conference.
- Participants have requested help for the conference (using 00 DTMF code) during the conference.
- Participants have problems connecting to conferences, for example, when they enter the wrong conference ID or password.

In addition, the RMX user (operator) can join the ongoing conference and assist all conference participants.

Operator assistance is available only when an *Operator conference* is running on the MCU.

The *Operator conference* offers additional conference management capabilities to the RMX users, enabling them to attend to participants with special requirements and acquire participant details for billing and statistics. This service is designed usually for large conferences that require the personal touch.

Operator assistance is available in both MPM and MPM+ *Card Configuration Modes*.

Operator Conferences

An *Operator conference* is a special conference that enables the RMX user acting as an operator to assist participants without disturbing the ongoing conferences and without being heard by other conference participants. The operator can move a participant from the Entry Queue or ongoing conference to a private, one-on-one conversation in the Operator conference.

In attended mode, the RMX user (operator) can perform one of the following actions:

- Participants connected to the Entry Queue who fail to enter the correct destination ID or conference password can be moved by the user to the Operator conference for assistance.
- After a short conversation, the operator can move the participant from the Operator conference to the appropriate destination conference (Home conference).
- The operator can connect participants belonging to the same destination conference to their conference simultaneously by selecting the appropriate participants and moving them to the Home conference (interactively or using the right-click menu).
- The operator can move one or several participants from an ongoing conference to the *Operator conference* for a private conversation.
- The operator can move participants between ongoing Continuous Presence conferences.

Operator Conference Guidelines

- An *Operator conference* can only run in Continuous Presence mode.
- *Operator conference* is defined in the Conference Profile. When enabled in Conference Profile, *High Definition Video Switching* option is disabled.
- An *Operator conference* can only be created by a User with Operator or Administrator *Authorization* level.
- *Operator conference* name is derived from the User Login Name and it cannot be modified.
- Only one *Operator conference* per User *Login Name* can be created.
- When created, the *Operator conference* must include one and only one participant - the Operator participant.
- Only a defined dial-out participant can be added to an *Operator conference* as an Operator participant
- Once running, the RMX user can add new participants or move participants from other conferences to this conference. The maximum number of participants in an *Operator conference* is the same as in standard conferences.
- Special icons are used to indicate an *Operator conference* in the Ongoing Conferences list and the operator participant in the Participants list.

- An *Operator conference* cannot be defined as a Reservation.
- An *Operator conference* can be saved to a Conference Template. An ongoing *Operator conference* can be started from a Conference Template.
- The Operator participant cannot be deleted from the *Operator conference*, but it can be disconnected from the conference.
- When deleting or terminating the *Operator conference*, the operator participant is automatically disconnected from the MCU, even if participating in a conference other than the *Operator conference*.
- Participants in Telepresence conferences cannot be moved from their conference, but an operator can join their conference and help them if assistance is required.
- Moving participants from/to an *Operator conference* follows the same guidelines as moving participants between conferences. For move guidelines, see "*Move Guidelines*" on page **1**.
- When a participant is moved from the Entry Queue to the *Operator conference*, the option to move back to the source (Home) conference is disabled as the Entry Queue is not considered as a source conference.
- The conference chairperson cannot be moved to the *Operator conference* following the individual help request if the *Auto Terminate When Chairperson Exits* option is enabled, to prevent the conference from automatically ending prematurely. In such a case, the assistance request is treated by the system as a conference assistance request, and the operator can join the conference.

Defining the Components Enabling Operator Assistance

To enable operator assistance for conferences, the following conferencing entities must be adjusted or created:

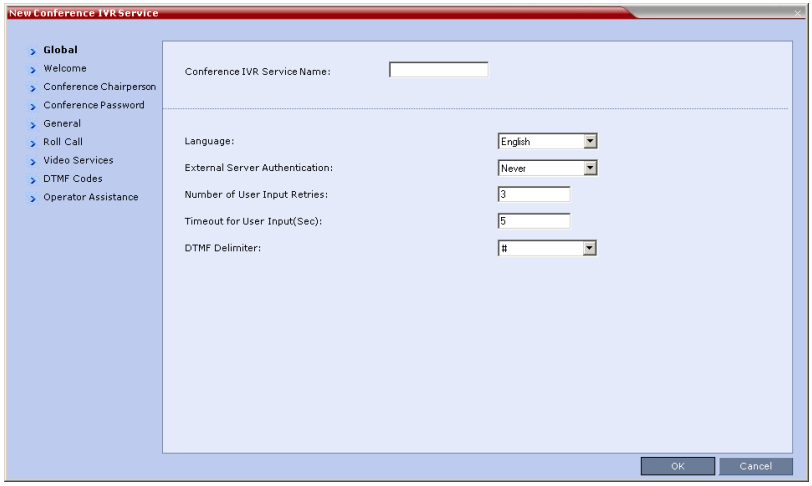
- IVR Service (Entry Queue and Conference) in which Operator Assistance options are enabled.
- A Conference Profile with the *Operator Conference* option enabled.
- An active Operator conference with a connected Operator participant.

Defining a Conference IVR Service with Operator Assistance Options

1 In the *RMX Management* pane, expand the *Rarely Used* list and click the **IVR Services** ( ) entry.

2 On the *IVR Services* toolbar, click the **New Conference IVR Service** ( ) button.

The *New Conference IVR Service - Global* dialog box opens.



3 Enter the *Conference IVR Service Name*.

4 Define the *Conference IVR Service - Global* parameters. For details, see the *RMX 2000 Administrator's Guide*, *Table 13-3, "Conference IVR Service Properties - Global Parameters,"* on page **13-9**.

5 Click the **Welcome** tab.
The *New Conference IVR Service - Welcome* dialog box opens.

6 Define the system behavior when the participant enters the Conference IVR queue. For details, see *RMX 2000 Administrator's Guide,* on page **13-11**.

7 Click the **Conference Chairperson** tab.
The *New Conference IVR Service - Conference Chairperson* dialog box opens.

8 If required, enable the chairperson functionality and select the various voice messages and options for the chairperson connection.

For details, see *RMX 2000 Administrator's Guide, Table 13-4, "New Conference IVR Service Properties - Conference Chairperson Options and Messages,"* on page **13-13**.

9 Click the **Conference Password** tab.

The *New Conference IVR Service - Conference Password* dialog box opens.

10 If required, enable the request for conference password before moving the participant from the conference IVR queue to the conference and set the MCU behavior for password request for *Dial-in* and *Dial-out* participant connections. For details, see *RMX 2000 Administrator's Guide, Table 13-5, "New Conference IVR Service Properties - Conference Password Parameters,"* on page **13-14**.

11 Select the various audio messages that will be played in each case. For details, see *RMX 2000 Administrator's Guide, Table 13-5, "New Conference IVR Service Properties - Conference Password Parameters,"* on page **13-14**.

12 Click the **General** tab.
The *New Conference IVR Service - General* dialog box opens.

13 Select the messages that will be played during the conference. For details, see *RMX 2000 Administrator's Guide, Table 13-6, "Conference IVR Service Properties - General Voice Messages,"* on page **13-16**.

14 Click the **Roll Call** tab.
The *New Conference IVR Service - Roll Call* dialog box opens.

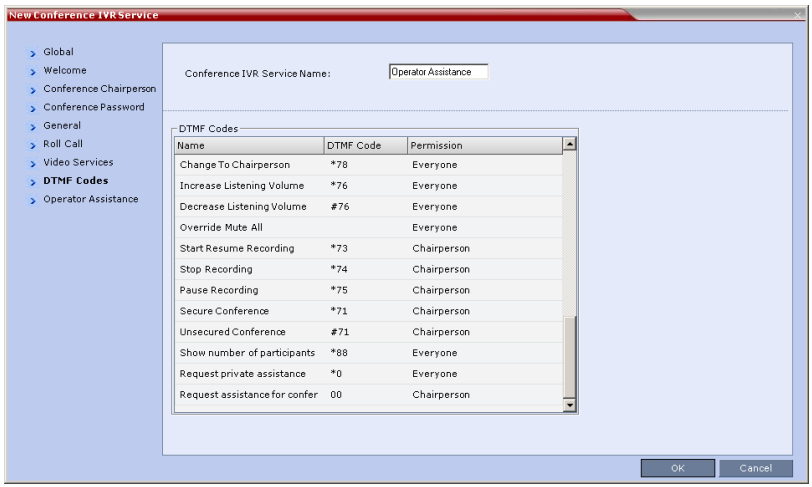
15 Enable the Roll Call feature and assign the appropriate audio file to each message type. For details, see *RMX 2000 Administrator's Guide, Table 13-7, "Conference IVR Service Properties - Roll Call Messages,"* on page **13-19**.

16 Click the **Video Services** tab.
The *New Conference IVR Service - Video Services* dialog box opens.

17 Define the *Video Services* parameters. For details, see *RMX 2000 Administrator's Guide, Table 13-8, "New Conference IVR Service Properties - Video Services Parameters,"* on page **13-21**.

18 Click the **DTMF Codes** tab.

The *New Conference IVR Service - DTMF Codes* dialog box opens.



The default DTMF codes for the various functions that can be performed during the conference by all participants or by the chairperson are listed. For the full list of the available DTMF codes, see *RMX 2000 Administrator's Guide, Table 13-9, "New Conference IVR Service Properties - DTMF Codes,"* on page **13-23**.

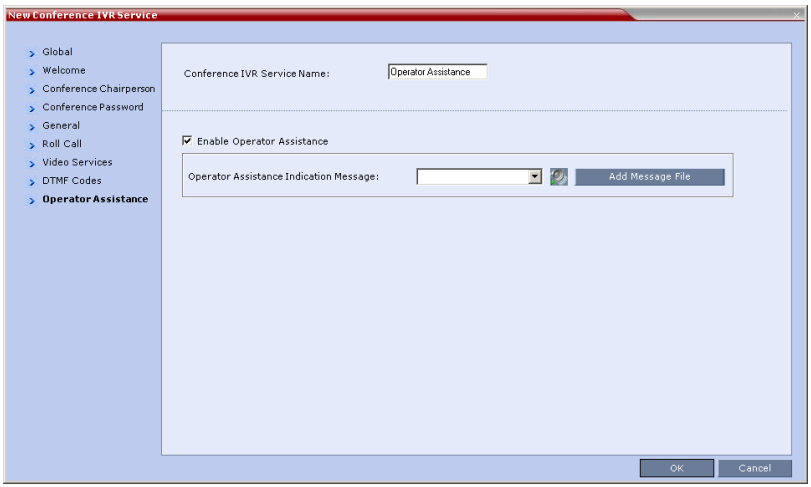
19 If required, modify the default DTMF codes and the permissions for various operations including Operator Assistance options:

— ***0** for individual help - the participant requested help for himself or herself. In such a case, the participant requesting help is moved to the Operator conference for one-on-one conversation. By default, all participants can use this code.

— **00** for conference help - the conference chairperson (default) can request help for the conference. In such a case, the operator joins the conference.

20 Click the **Operator Assistance** tab.

The *Operator Assistance* dialog box opens.



21 Select **Enable Operator Assistance** to enable operator assistance when the participant requires or requests help during the connection process to the conference or during the conference.

22 In the *Operator Assistance Indication Message* field, select the audio message to be played when the participant requests or is waiting for the operator's assistance.

If the audio file was not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the RMX.

23 Click **OK** to complete the IVR Service definition.
The new Conference IVR Service is added to the *IVR Services* list.

Defining an Entry Queue IVR Service with Operator Assistance Options

1 In the *RMX Management* pane, click **IVR Services** ( ).

2 In the *IVR Services* list, click the **New Entry Queue IVR Service** ( ) button.

The *New Entry Queue IVR Service - Global* dialog box opens.

3 Define the *Entry Queue Service Name*.

4 Define the Entry Queue IVR Service Global parameters. For details, see *RMX 2000 Administrator's Guide*, *Table 13-10, "Entry Queue IVR Service Properties - Global Parameters,"* on page **13-27**.

5 Click the **Welcome** tab.
The *New Entry Queue IVR Service - Welcome* dialog box opens.

6 Define the system behavior when the participant enters the Entry Queue. This dialog box contains options that are identical to those in the *Conference IVR Service - Welcome Message* dialog box. For details, see *RMX 2000 Administrator's Guide, on page* **13-11**.

7 Click the **Conference ID** tab.
The *New Entry Queue IVR Service - Conference ID* dialog box opens.

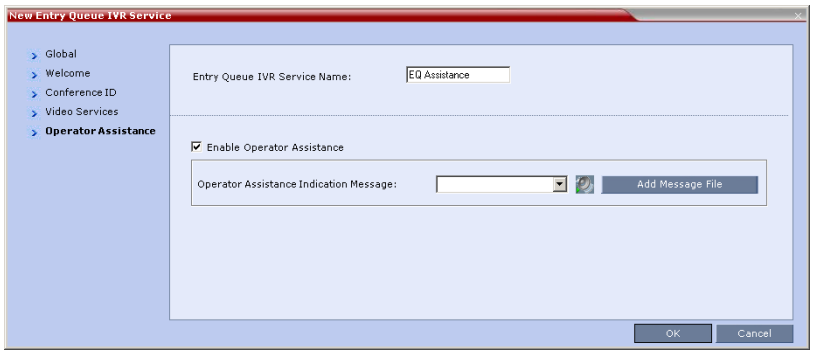
8 Select the required voice messages. For detail, see *RMX 2000 Administrator's Guide, Table 13-11, "Entry Queue IVR Service Properties - Conference ID,"* on page **13-29**.

9 Click the **Video Services** tab.
The *New Entry Queue IVR Service - Video Services* dialog box opens.

10 In the *Video Welcome Slide* list, select the video slide that will be displayed to participants connecting to the Entry Queue. The slide list includes the video slides that were previously uploaded to the MCU memory.

11 Click the **Operator Assistance** tab.

The *Operator Assistance* dialog box opens.



12 Select **Enable Operator Assistance** to enable operator assistance when the participant requires or requests help during the connection process.

13 In the *Operator Assistance Indication Message* field, select the audio message to be played when the participant requests or is waiting for operator's assistance.

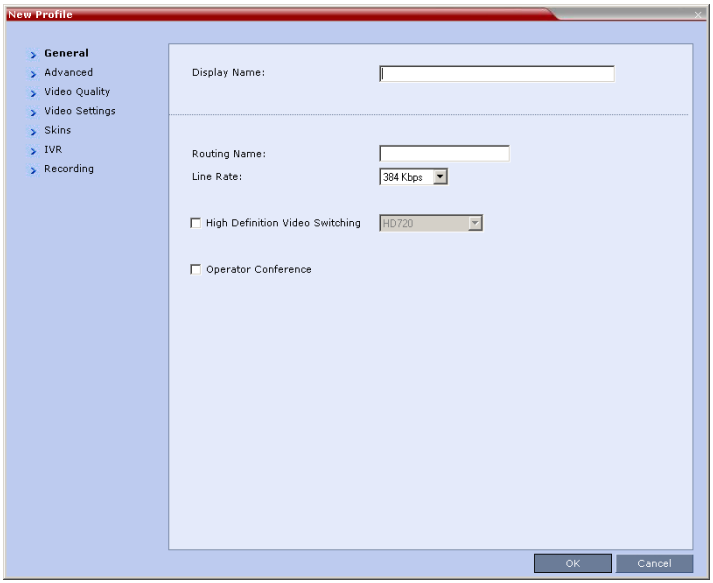
If the audio file was not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the RMX.

14 Click **OK** to complete the Entry Queue IVR Service definition. The new Entry Queue IVR Service is added to the *IVR Services* list.

Defining a Conference Profile for an Operator Conference

1 In the *RMX Management* pane, click **Conference Profiles**.

2 In the *Conference Profiles* pane, click the **New Profile** button. The *New Profile – General* dialog box opens.



3 Define the Profile name and, if required, the Profile general parameters:

Table 7-1 *New Profile - General Parameters*

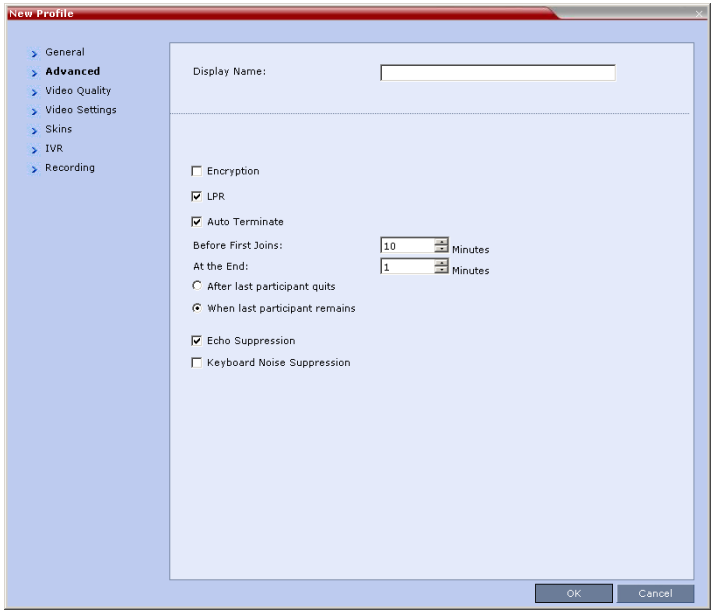
Field/Option	Description
Display Name	Enter a unique Profile name, as follows: • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. It is recommended to use a name that indicates the Profile type, such as Operator conference or Video Switching conference. **Note:** This is the only parameter that must be defined when creating a new profile.
Routing Name	Enter the Profile name using ASCII characters set. The Routing Name can be defined by the user or automatically generated by the system if no Routing Name is entered as follows: • If an all ASCII text is entered in Display Name, it is used also as the Routing Name. • If any combination of Unicode and ASCII text (or full Unicode text) is entered in Display Name, the ID (such as Conference ID) is used as the Routing Name.
Line Rate	Select the conference bit rate. The line rate represents the combined video, audio and Content rate. The default setting is 384 Kbps.

Table 7-1 *New Profile - General Parameters (Continued)*

Field/Option	Description
High Definition Video Switching	If the *Operator Conference* option is selected, this option is disabled, and the selection is cleared. When selected, the conference is ultra-high quality video resolution, in a special conferencing mode which implies that all participants must connect at the same line rate and use HD video. This feature utilizes the resources more wisely and efficiently by: • Saving utilization of video ports (1 port per participant as opposed to 4 ports in CP mode). • Video display is in full screen mode only and video is switched to the speaker. Drawbacks of this feature are that all participants must connect at the same line rate, (e.g. HD) and all participants with endpoints not supporting HD will connect as secondary (audio only). Select the High Definition resolution; select either HD 720p or HD 1080p (in MPM+ mode only). If HD 1080p is selected, endpoints that do not support HD 1080p resolution are connected as Secondary (Audio Only) participants. **Notes:** • High Definition Video Switching conferencing mode is unavailable to ISDN participants. For more information, see *RMX 2000 Administrator's Guide*, "*Video Resolutions in CP"* on page **2-3**.
Operator Conference	Select this option to define the profile of an Operator conference. An Operator conference can only be a Continuous Presence conference, therefore when selected, the *High Definition Video Switching* option is disabled and cleared. When defining an *Operator Conference*, the *Send Content to Legacy Endpoints* option in the *Video Settings* tab is cleared and disabled.

4 Click the **Advanced** tab.

The *New Profile – Advanced* dialog box opens.



5 Define the following parameters:

Table 7-2 *New Profile - Advanced Parameters*

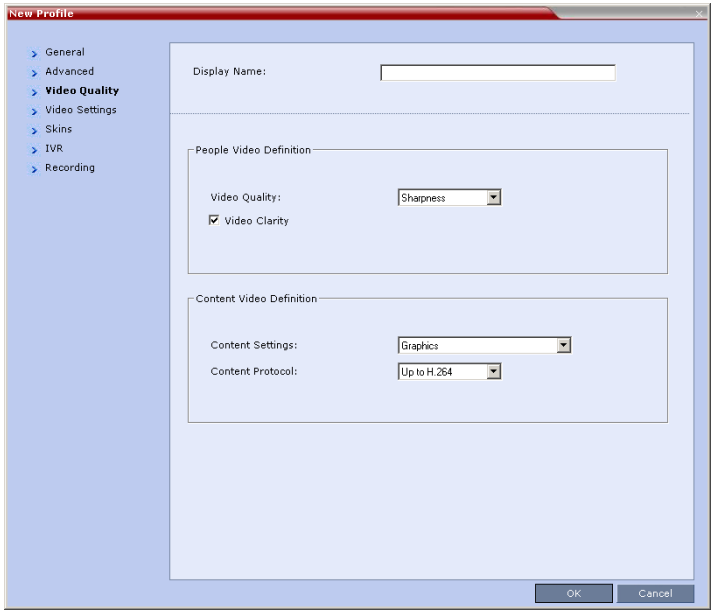
Field/Option	Description
Encryption	Select this check box to activate encryption for the conference. For more information, see *RMX 2000 Administrator's Guide*, "*Media Encryption"* on page **2-30**.
LPR	When selected (default for CP conferences), *Lost Packet Recovery* creates additional packets that contain recovery information used to reconstruct packets that are lost during transmission. LPR is automatically disabled if High Definition Video Switching is selected. For more information, see *RMX 2000 Administrator's Guide, "LPR – Lost Packet Recovery"* on page **2-36**.

Table 7-2 *New Profile - Advanced Parameters (Continued)*

Field/Option	Description
Auto Terminate	When selected (default), the conference automatically ends when the termination conditions are met: ***Before First Joins*** — No participant has connected to a conference during the *n* minutes after it started. Default idle time is 10 minutes. ***At the End - After Last Quits*** — All the participants have disconnected from the conference and the conference is idle (empty) for the predefined time period. Default idle time is 1 minute. ***At the End - When Last Participant Remains*** — Only one participant is still connected to the conference for the predefined time period (excluding the recording link which is not considered a participant when this option is selected). This option should be selected when defining a Profile that will be used for Gateway Calls and you want to ensure that the call is automatically terminated when only one participant is connected. Default idle time is 1 minute. **Note:** The selection of this option is automatically cleared and disabled when the *Operator Conference* option is selected. The Operator conference cannot automatically end unless it is terminated by the RMX User.
Echo Suppression	When enabled (default), an algorithm is used to search for and detect echo outside the normal range of human speech (such as echo) and automatically mute them when detected. Clear this option to disable the Echo Suppression algorithm. **Note:** This option is activated only in *MPM+ Card Configuration Mode*.
Keyboard Noise Suppression	When enabled, an algorithm is used to search for and detect keyboard noises and automatically mute them when detected. **Note:** This option is activated only in *MPM+ Card Configuration Mode*.

6 Click the **Video Quality** tab.

The *New Profile – Video Quality* dialog box opens.



7 Define the following parameters:

Table 7-3 *New Profile - Video Quality Parameters*

Field/Option	Description
People Video Definition	
Video Quality	Depending on the amount of movement contained in the conference video, select either: • ***Motion*** – for a higher frame rate without increased resolution • ***Sharpness*** – for higher video resolution and requires more system resources **Note:** When Sharpness is selected as the Video Quality setting in the conference Profile, the RMX will send 4CIF (H.263) at 15fps instead of CIF (H.264) at 30fps. For more information, see *RMX 2000 Administrator's Guide*, "*Video Resolutions in CP"* on page **2-3**.

Table 7-3 *New Profile - Video Quality Parameters (Continued)*

Field/Option	Description
Video Clarity™	When enabled (default), *Video Clarity* applies video enhancing algorithms to incoming video streams of resolutions up to and including SD. Clearer images with sharper edges and higher contrast are sent back to all endpoints at the highest possible resolution supported by each endpoint. All layouts, including 1x1, are supported. Video Clarity can only be enabled for Continuous Presence conferences in MPM+ Card Configuration Mode.
Content Video Definition	
Content Settings	Select the transmission mode for the Content channel: • ***Graphics*** — basic mode, intended for normal graphics • ***Hi-res Graphics*** — a higher bit rate intended for high resolution graphic display • ***Live Video*** — Content channel displays live video Selection of a higher bit rate for the Content results in a lower bit rate for the people channel. For more information, see *RMX 2000 Administrator's Guide*, "*H.239"* on page **2-12**.
Content Protocol	**H.263** – Content is shared using *H.263* even if some endpoints have *H.264* capability. **Up to H.264** – *H.264* is the default Content sharing algorithm. When selected: • Content is shared using *H.264* if all endpoints have *H.264* capability. • Content is shared using *H.263* if all endpoints do not have *H.264* capability. • Endpoints that do not have at least *H.263* capability can connect to the conference but cannot share Content.

8 Click the **Video Settings** tab.

The *New Profile - Video Settings* dialog box opens.

9 Define the video display mode and layout. For more details, see *RMX 2000 Administrator's Guide, Table 1-7, "Profile Properties - Video Settings,"* on page **1-16**.

10 Click the **Skins** tab to modify the background and frames.
The *New Profile - Skins* dialog box opens.

11 Select one of the *Skin* options.

12 Click **IVR** tab.

The *New Profile - IVR* dialog box opens.

13 Select the IVR Service and if the conference requires a chairperson.

14 **Optional**. Click the **Recording** tab to enable conference recording with *Polycom RSS 2000*.

15 Define the various recording parameters. for details, see *RMX 2000 Administrator's Guide, Table 1-11, "Profile Properties - Recording Parameters,"* on page **1-23**.

16 Click **OK** to complete the *Profile* definition.
A new *Profile* is created and added to the *Conference Profiles* list.

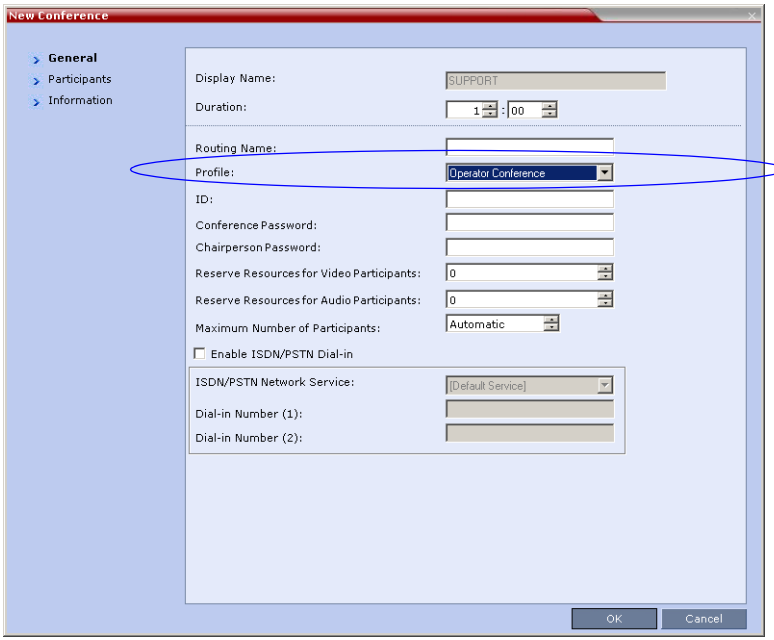
Defining an Ongoing Operator Conference

To start a conference from the Conference pane:

1 In the *Conferences* pane, click the **New Conference** ( ) button.

The *New Conference – General* dialog box opens.

2 In the *Profile* field, select a Profile in which the *Operator Conference* option is selected.



Upon selection of the Operator Conference Profile, the *Display Name* is automatically taken from the RMX User *Login Name*. This name cannot be modified.

Only one Operator conference can be created for each User Login name.

3 Define the following parameters:

Table 7-4 *New Conference – General Options*

Field	Description
Duration	Define the duration of the conference in hours using the format HH:MM (default 01:00). **Notes:** • The Operator conference is automatically extended up to a maximum of 168 hours. Therefore, the default duration can be used. • This field is displayed in all tabs.

Table 7-4 *New Conference – General Options (Continued)*

Field	Description
Routing Name	*Routing Name* is the name with which ongoing conferences, Meeting Rooms, Entry Queues and SIP Factories register with various devices on the network such as gatekeepers and SIP server. This name must defined using ASCII characters. **Comma, colon and semicolon characters cannot be used in the *Routing Name*.** The *Routing Name* can be defined by the user or automatically generated by the system if no *Routing Name* is entered as follows: • If ASCII characters are entered as the *Display Name*, it is used also as the *Routing Name* • If a combination of Unicode and ASCII characters (or full Unicode text) is entered as the *Display Name*, the *ID* (such as Conference ID) is used as the *Routing Name.* If the same name is already used by another conference, Meeting Room or Entry Queue, the RMX displays an error message and requests that you to enter a different name.
ID	Enter the unique-per-MCU conference ID. If left blank, the MCU automatically assigns a number once the conference is launched. This ID must be communicated to conference participants to enable them to dial in to the conference.
Conference Password	Leave this field empty when defining an Operator conference.
Chairperson Password	Leave this field empty when defining an Operator conference.

Table 7-4 *New Conference – General Options (Continued)*

Field	Description
Reserve Resources for Video Participants	Enter the number of video participants for which the system must reserve resources. Default: 0 participants. When defining an Operator conference it is recommended to reserve resources for at least 2 video participants (for the operator and one additional participant - who will be moved to the Operator conference for assistance). Maximum: • MPM Mode: 80 participants. • MPM+ Mode: 80 participants.
Reserve Resources for Audio Participants	Enter the number of audio participants for which the system must reserve resources. Default: 0 participants. When defining an Operator conference and the operator is expected to help voice participants, it is recommended to reserve resources for at least 2 video participants (for the operator and one additional participant - who will be moved to the Operator conference for assistance). Maximum: • MPM Mode: 80 participants. • MPM+ Mode: 200 audio participants.
Maximum Number of Participants	Enter the maximum number of participants that can connect to an Operator conference (you can have more than two), or leave the default selection (Automatic). Maximum number of participants that can connect to an Operator conference: • MPM Mode: 80 participants. • MPM+ Mode: 200 participants (80 video and 120 audio or 200 audio)

Table 7-4 *New Conference – General Options (Continued)*

Field	Description
Enable ISDN/ PSTN Dial-in	Select this check box if you want ISDN and PSTN participants to be able to connect directly to the Operator conference. This may be useful if participants are having problems connecting to their conference and you want to identify the problem or help them connect to their destination conference.
ISDN/PSTN Network Service and Dial-in Number	If you have enable the option for ISDN/PSTN direct dial-in to the Operator conference, assign the ISDN/ PSTN Network Service and a dial-in number to be used by the participants, or leave these fields blank to let the system select the default Network Service and assign the dial-in Number. **Note:** The dial-in number must be unique and it cannot be used by any other conferencing entity.

4 Click the **Participants** tab.
The *New Conference - Participants* dialog box opens.

You must define or add the Operator participant to the Operator conference.

This participant must be defined as a **dial-out** participant.

Define the parameters of the endpoint that will be used by the RMX User to connect to the Operator conference and to other conference to assist participants.

For more details, see the *RMX 2000 Getting Started Guide*, *"Participants Tab"* on page **3-21**.

5 **Optional**. Click the **Information** tab**.**
The *Information* tab opens.

6 Enter the required information. For more details, see the *RMX 2000 Getting Started Guide*, *"Information Tab"* on page **3-25**.

7 Click **OK**.

The new Operator conference is added to the ongoing *Conferences* list with a special icon .

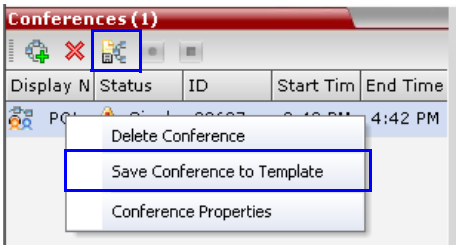
The Operator participant is displayed in the *Participants* list with an Operator participant icon , and the system automatically dials out to the Operator participant.

Saving an Operator Conference to a Template

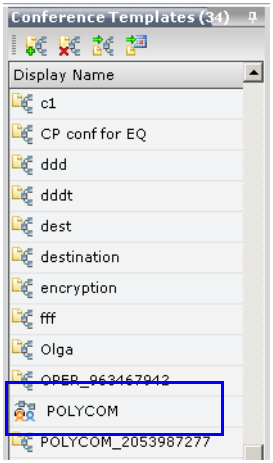
The Operator conference that is ongoing can be saved as a template.

To save an ongoing Operator conference as a template:

1. In the *Conferences List*, select the Operator conference you want to save as a Template.
2. Click the **Save Conference to Template** ( ) button.
   or
   Right-click and select **Save Conference to Template**.



The conference is saved to a template whose name is taken from the ongoing conference *Display Name* (the Login name of the RMX User). The Template appears with the Operator Conference icon.



Starting an Operator Conference from a Template

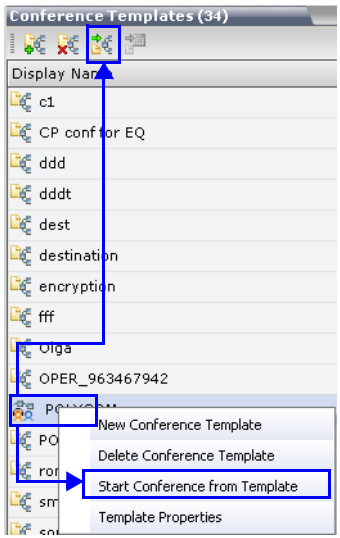
An ongoing Operator conference can be started from an Operator Template saved in the *Conference Templates* list.

To start an ongoing Operator conference from an Operator Template:

1 In the *Conference Templates* list, select the Operator Template to start as an ongoing Operator conference.

- You can only start an Operator conference from a template whose name is identical to your Login Name. For example, if your Login name is Polycom, you can only start an Operator conference from a template whose name is Polycom.
- If an ongoing Operator conference with the same name or any other conference with the same ID is already running, you cannot start another Operator conference with the same login name.

2 Click the **Start Conference from Template** ( ) button.
or
Right-click and select **Start Conference from Template**.



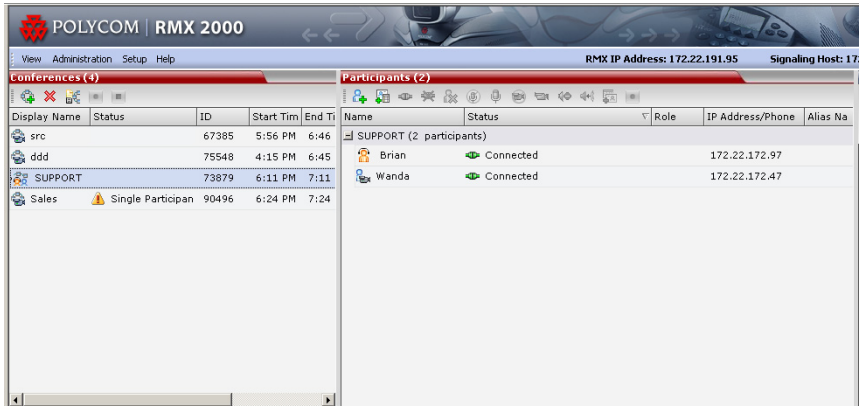
The conference is started.

The name of the ongoing conference in the *Conferences* list is taken from the Conference Template *Display Name*.

Monitoring Operator Conferences and Participants Requiring Assistance

Operator conferences are monitored in the same way as standard ongoing conferences.

Each Operator conference includes at least one participant - the Operator.


POLYCOM | RMX 2000
View Administration Setup Help
RMX IP Address: 172.22.191.95
Signaling Host: 17
Conferences (4)
Display Name Status ID Start Tim End Ti
src 67385 5:56 PM 6:46
ddd 75548 4:15 PM 6:45
SUPPORT 73879 6:11 PM 7:11
Sales Single Participan 90496 6:24 PM 7:24
Participants (2)
Name Status Role IP Address/Phone Alias Na
SUPPORT (2 participants)
Brian Connected 172.22.172.97
Wanda Connected 172.22.172.47

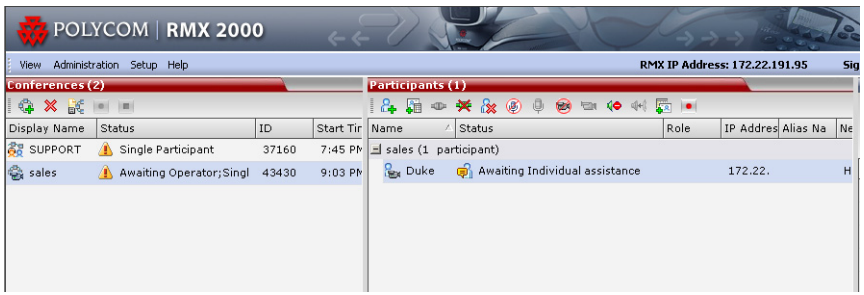
You can view the properties of the *Operator conference* by double-clicking the conference entry in the *Conferences* list or by right-clicking the conference entry and selecting **Conference Properties**. For more information, see the *RMX 2000 Getting Started Guide*, "*Conference Level Monitoring*" on page **3-43**.

Requesting Help

A participant can request help using the appropriate DTMF code from his/her touch tone telephone or the endpoint's DTMF input device. The participant can request *Individual Assistance* (default DTMF code ***0**) or *Conference Assistance* (default DTMF code **00**).

Participants in Entry Queues who failed to enter the correct destination conference ID or the conference password will wait for operator assistance (provided that an Operator conference is active).

When requiring or requesting operator assistance, the RMX management application displays the following:


POLYCOM | RMX 2000
View Administration Setup Help
RMX IP Address: 172.22.191.95
Conferences (2)
Participants (1)
Display Name Status ID Start Tir
SUPPORT Single Participant 37160 7:45 PM
sales Awaiting Operator;Singl 43430 9:03 PM
Name Status Role IP Addres Alias Na
sales (1 participant)
Duke Awaiting Individual assistance 172.22.

- The participant's connection *Status* changes, reflecting the help request. For details, see Table 1-5.
- The conference status changes and it appears with the exclamation point icon and the status "Awaiting Operator".
- The appropriate voice message is played to the relevant participants indicating that assistance will be provided shortly.

The following icons and statuses are displayed in the *Participant Status* column:

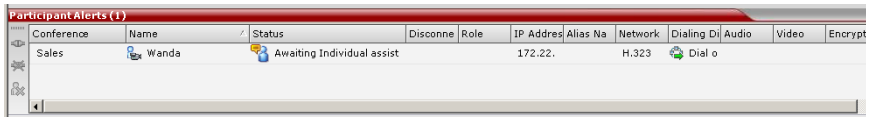
Table 7-5 *Participants List Status Column Icons and Indications*

Icon	Status indication	Description
	Awaiting Individual Assistance	The participant has requested the operator's assistance for himself/ herself.
	Awaiting Conference Assistance	The participant has requested the operator's assistance for the conference. Usually this means that the operator is requested to join the conference.

When the Operator moves the participant to the *Operator conference* for individual assistance the participant Status indications are cleared.

Participant Alerts List

The *Participant Alerts* list contains all the participants who are currently waiting for operator assistance.



Participants are automatically added to the *Participants Alerts* list in the following circumstances:

- The participant fails to connect to the conference by entering the wrong conference ID or conference password and waits for the operator's assistance
- The participant requests Operator's Assistance during the ongoing conference

This list is used as reference only. Participants can be assisted and moved to the *Operator conference* or the destination conference only from the *Participants* list of the Entry Queues or ongoing conference where they are awaiting assistance.

The participants are automatically removed from the *Participant Alerts* list when moved to any conference (including the *Operator conference*).

Moving Participants Between Conferences

The RMX User can move participants between ongoing conferences, including the *Operator conference*, and from the Entry Queue to the destination conference if help is required.

When moving between conferences or when a participant is moved from an Entry Queue to a conference by the RMX user (after failure to enter the correct destination ID or conference password), the IVR messages and slide display are skipped.

Move Guidelines

- Move is available only between CP conferences. Move is unavailable from/to Video Switching conferences.
- Move between conferences can be performed without an active *Operator conference*.
- When moving the conference chairperson from his/her conference to another conference, the source conference will automatically end if

the *Auto Terminate When Chairperson Exits* option is enabled and that participant is the only conference chairperson.

- When moving the Operator to any conference (following assistance request), the IVR messages and slide display are skipped.
- Participants cannot be moved from a Telepresence conference.
- Participants cannot be moved from LPR-enabled conferences to non-LPR conferences. Move from non-LPR conferences to LPR-enabled conferences is available.
- Move between encrypted and non-encrypted conferences depends on the **ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF** flag setting, as described in Table 1-6:

Table 7-6 *Participant Move Capabilities vs. ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF flag setting*

Flag Setting	Source Conference/ EQ Encrypted	Destination Conference Encrypted	Move Enabled?
NO	Yes	Yes	Yes
NO	Yes	No	Yes
NO	No	Yes	No
NO	No	No	Yes
YES	Yes	Yes	Yes
YES	Yes	No	Yes
YES	No	Yes	Yes
YES	No	No	Yes

- When moving dial-out participants who are disconnected to another conference, the system automatically dials out to connect them to the destination conference.
- Cascaded links cannot be moved between conferences.
- Participants cannot be moved to a conference if the move will cause the number of participants to exceed the maximum number of participants allowed for the destination conference.

Moving Participants

RMX users can assist participants by performing the following operations:

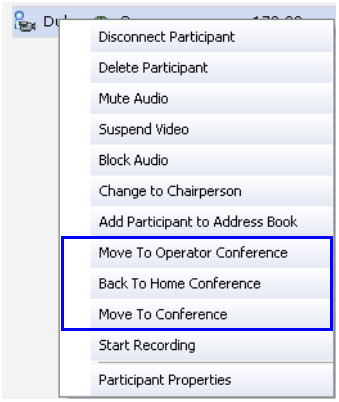
- Move a participant to an *Operator conference* (Attend a participant).
- Move a participant to the Home (destination) conference.
- Move participant from one ongoing conference to another

A move can be performed using the following methods:

- Using the participant right-click menu
- Using drag and drop

To move a participant from the ongoing conference using the right-click menu options:

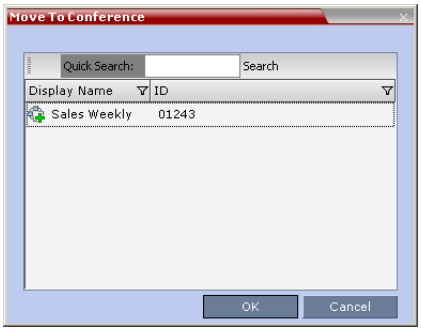
1. In the *Conferences* list, click the conference where there are participants waiting for Operator's Assistance to display the list of participants.
2. In the *Participants* list, right-click the icon of the participant to move and select one of the following options:



— **Move to Operator Conference** - to move the participant to the Operator conference

— **Move to Conference** - to move the participant to any ongoing conference.

When selected, the *Move to Conference* dialog box opens, letting you select the name of the destination conference.



— **Back to Home Conference** - if the participant was moved to another conference or to the *Operator conference*, this options moves the participant back to his/her source conference.

This option is not available if the participant was moved from the Entry Queue to the *Operator conference* or the destination conference.

Moving a Participant Interactively

You can drag and drop a participant from the Entry Queue or ongoing conference to the Operator or destination (Home) conference:

1. Display the participants list of the Entry Queue or the source conference by clicking its entry in the *Conferences* list.
2. In the Participants list, drag the icon of the participant to the *Conferences List* pane and drop it on the *Operator Conference* icon or another ongoing conference.

8

Conference Templates

Conference Templates enable administrators and operators to create, save, schedule and activate identical conferences.

A *Conference Template*:

- Saves the conference Profile.
- Saves all participant parameters including their *Personal Layout* and *Video Forcing* settings.
- Simplifies the setting up *Telepresence* conferences where precise participant layout and video forcing settings are crucial.

Guidelines

- A maximum of 100 *Conference Templates* can be saved.
- A maximum of 200 participants can be saved in a *Conference Template* when the RMX is in MPM+ mode. When the RMX is in MPM mode, the maximum is 80 participants.
- If the RMX is switched to from MPM+ mode to MPM mode, conference templates may include more participants than the allowed maximum in MPM mode.

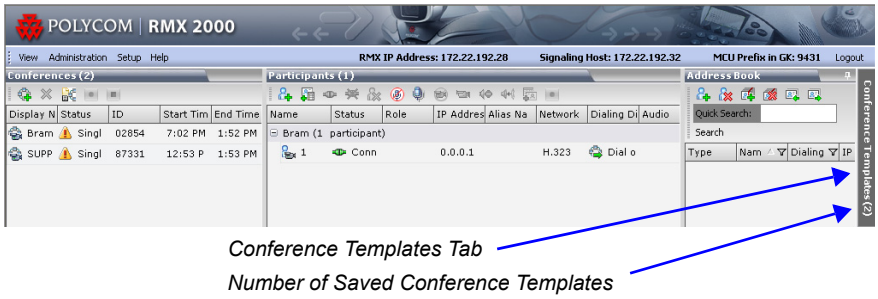
  Trying to start a *Conference Template* that exceeds the allowed maximum number of participants will result in participants being disconnected due to resource deficiency.
- If the Profile assigned to a conference is deleted while the conference is ongoing the conference cannot be saved as a template.
- A Profile assigned to a *Conference Template* cannot be deleted. The system does not permit such a deletion.
- Profile parameters are not embedded in the *Conference Template*, and are taken from the Profile when the *Conference Template* becomes an ongoing conference. Therefore, any changes to the Profile parameters between the time the *Conference Template* was created and the time

that it is activated (and becomes an ongoing conference) will be applied to the conference.

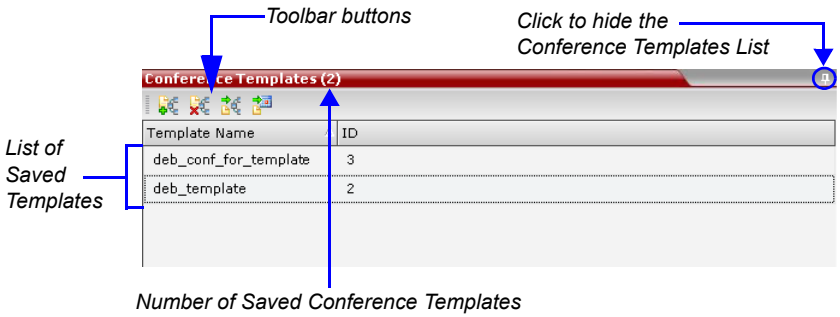
- Only defined participants can be saved to the *Conference Template*. Before saving a conference to a template ensure that all undefined participants have disconnected.
- Undefined participants are not saved in *Conference Templates*.
- Participant properties are embedded in the *Conference Template* and therefore, if the participant properties are modified in the Address Book after the *Conference Template* has been created they are not applied to the participant whether the *Template* becomes an ongoing conference or not.
- The *Conference Template* display name, routing name or ID can be the same as an Ongoing Conference, reservation, Meeting Room or Entry Queue as it is not active. However, an ongoing conference cannot be launched from the *Conference Template* if an ongoing conference, Meeting Room or Entry Queue already has the same name or ID. Therefore, it is recommended to modify the template ID, display name, routing name to be unique.
- A *Reservation* that has become an ongoing conference can be saved as *Conference Template*.
- SIP Factories and Entry Queues cannot be saved as *Conference Templates*.

Using Conference Templates

The *Conference Templates* list is initially displayed as a closed tab in the *RMX Web Client* main window. The number of saved *Conference Templates* is indicated on the tab.



Clicking the tab opens the *Conference Templates* list.



The *Conference Templates* are listed by *Conference Template Display Name* and *ID* and can be sorted by either field. The list can be customized by re-sizing the pane, adjusting the column widths or changing the order of the column headings.

For more information see *RMX 2000 Getting Started Guide*, "*Customizing the Main Screen*" on page **3-12**.

Clicking the anchor pin ( ) button hides the *Conference Templates* list as a closed tab.

Toolbar Buttons

The *Conference Template* toolbar includes the following buttons:

Table 1 *Conference Templates – Toolbar Buttons*

Button	Description
New Conference Template	Creates a new Conference Template.
Delete Conference Template	Deletes the Conference Template(s) that are selected in the list.
Start Conference from Template	Starts an ongoing conference from the *Conference Template* that has an identical name, ID parameters and participants as the template.
Schedule Reservation from Template	Creates a conference Reservation from the Conference Template with the same name, ID, parameters and participants as the Template. Opens the *Scheduler* dialog box enabling you to modify the fields required to create a single or recurring *Reservation* based on the template. For more information see "*Reservations"* on page **6-1**.

The ***Conferences List*** toolbar includes the following button:

Table 2 *Conferences List – Toolbar Button*

Button	Description
Save Conference to Template	Saves the selected ongoing conference as a Conference Template.

Creating a New Conference Template

There are two methods to create a *Conference Template*:

- From scratch - defining the conference parameters and participants
- Saving an ongoing conference as Template

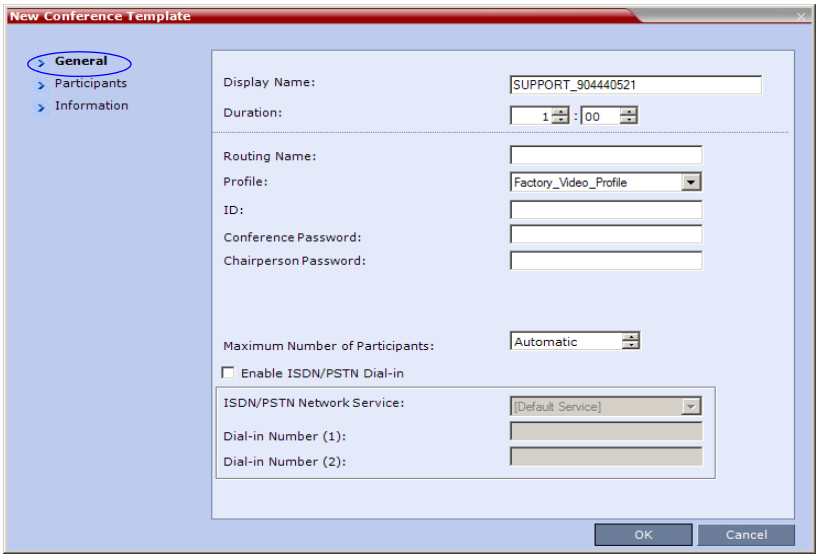
Creating a new Conference Template from Scratch

To create a new Conference Template:

1 In the *RMX Web Client*, click the **Conference Templates** tab.

2 Click the **New Conference Template** ( ) button.

The *New Conference Template - General* dialog box opens.



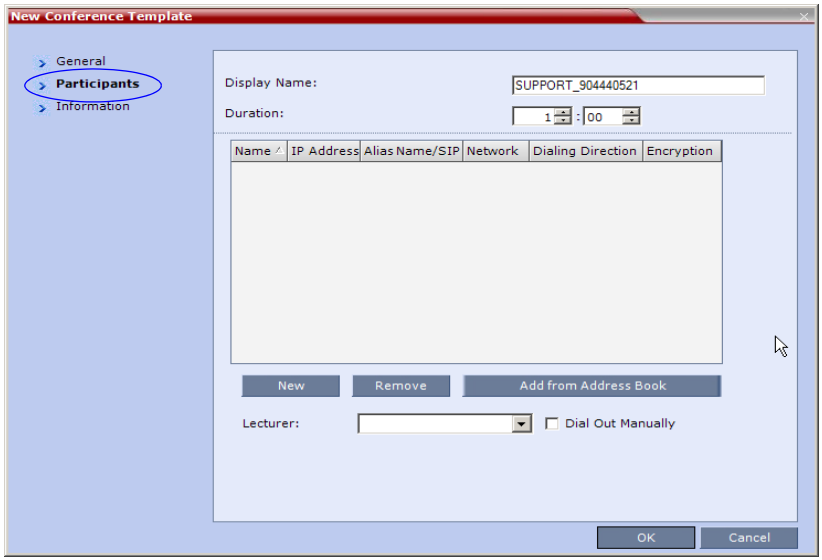
3 The fields of the *New Template – General* dialog box are identical to those of the *New Conference – General* dialog box. For a full description of the fields see the *RMX 2000 Getting Started Guide, "General Tab"* on page **3-17**.

4 Modify the fields of the *General* tab.

A unique dial-in number must be assigned to each conferencing entity. However, Conference Templates can be assigned dial-in numbers that are already assigned to other conferencing entities, but when the template is used to start an ongoing conference or schedule a reservation, it will not start if another ongoing conference, Meeting Room, Entry Queue or Gateway Profile is using this number.

5 Click the **Participants** tab.

The *New Template – Participants* dialog box opens.



The fields of the *New Template – Participants* dialog box are the same as those of the *New Conference – Participant* dialog box.

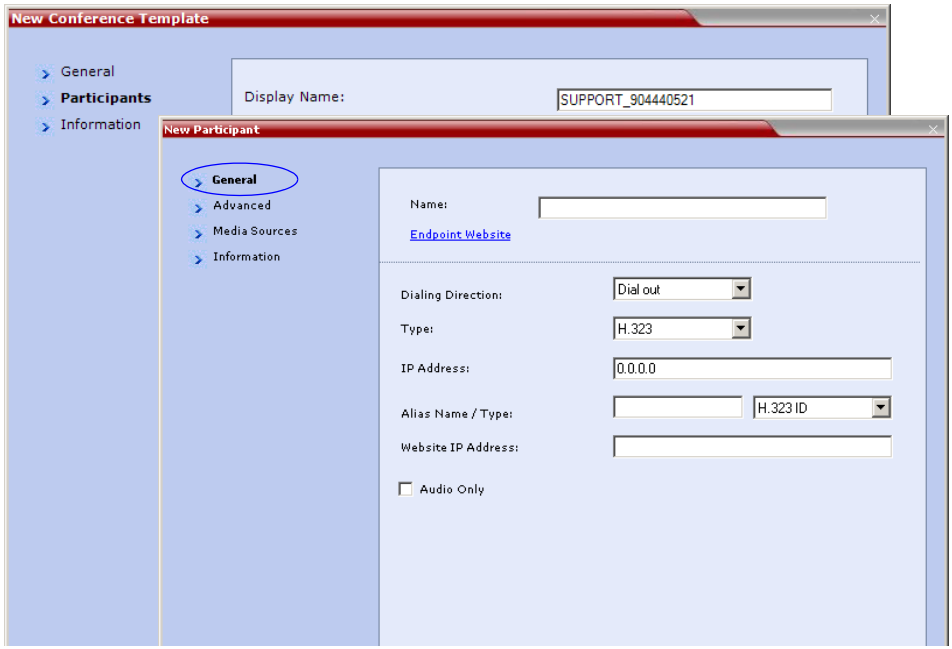
For a full description of these fields see the *RMX 2000 Getting Started Guide*, "*Participants Tab*" on page **3-21**.

6 **Optional.** Add participants to the template from the *Address Book*.

7 Click the **New** button.

The *New Participant – General* tab opens.

The *New Template – Participant* dialog box remains open in the background.

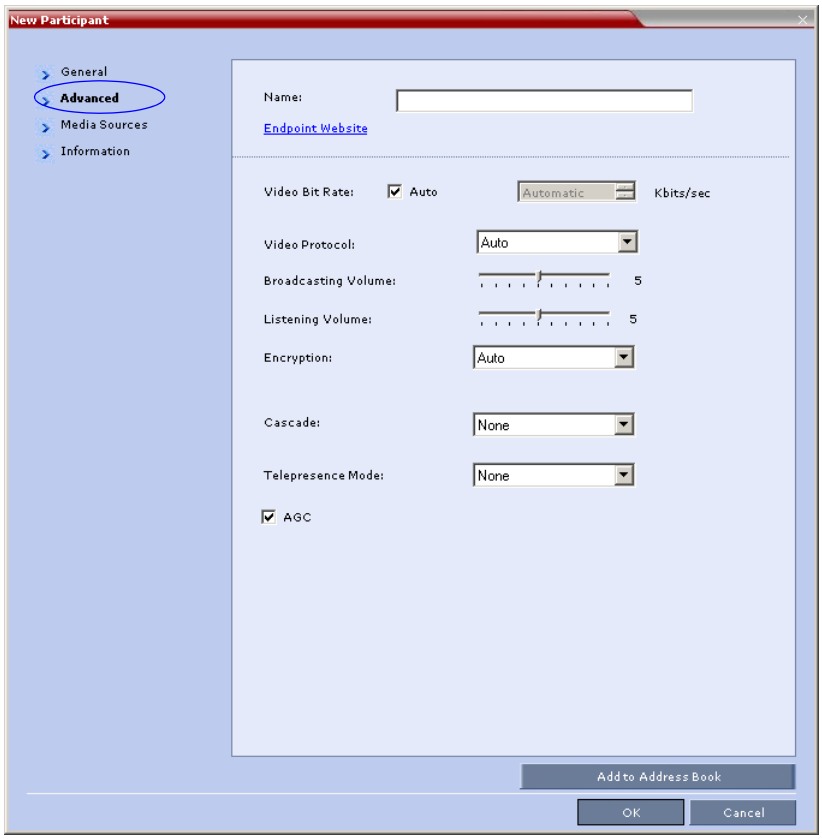


For a full description of the *General* tab fields see the *RMX 2000 Administrator's Guide*, "*Adding a new participant to the Address Book Directly*" on page 5-4.

8 Modify the fields of the *General* tab.

9 Click the **Advanced** tab.

The *New Participant – Advanced* tab opens.

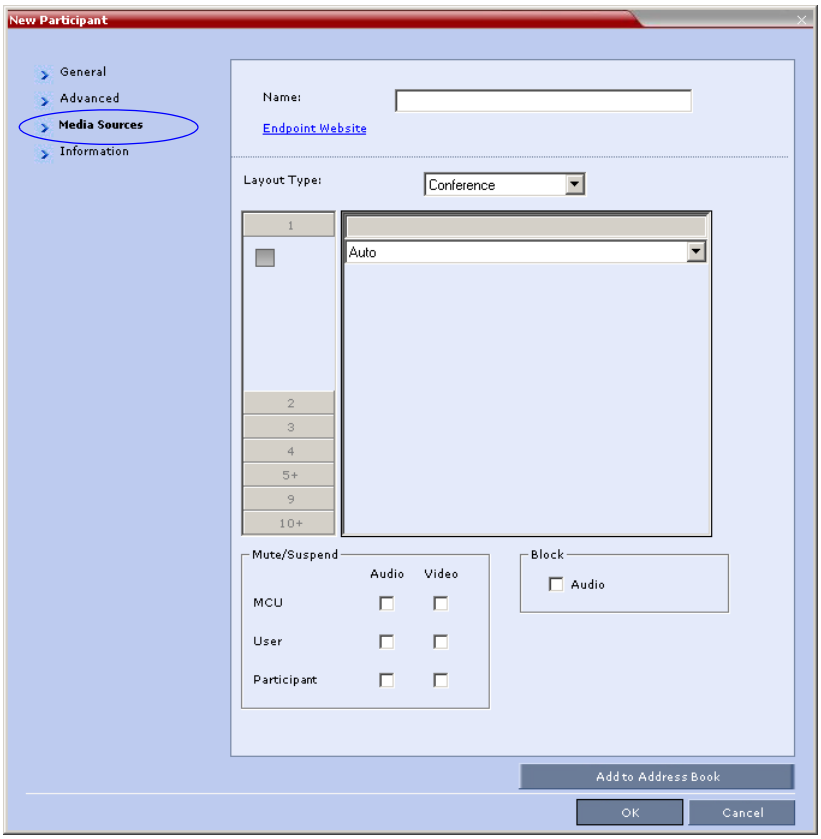


For a full description of the *Advanced* tab fields see the *RMX 2000 Administrator's Guide*, "*New Participant – Advanced Properties*" on page 5-9.

10 Modify the fields of the *Advanced* tab.

11 Click the **Media Sources** tab.

The *Media Sources* tab opens.



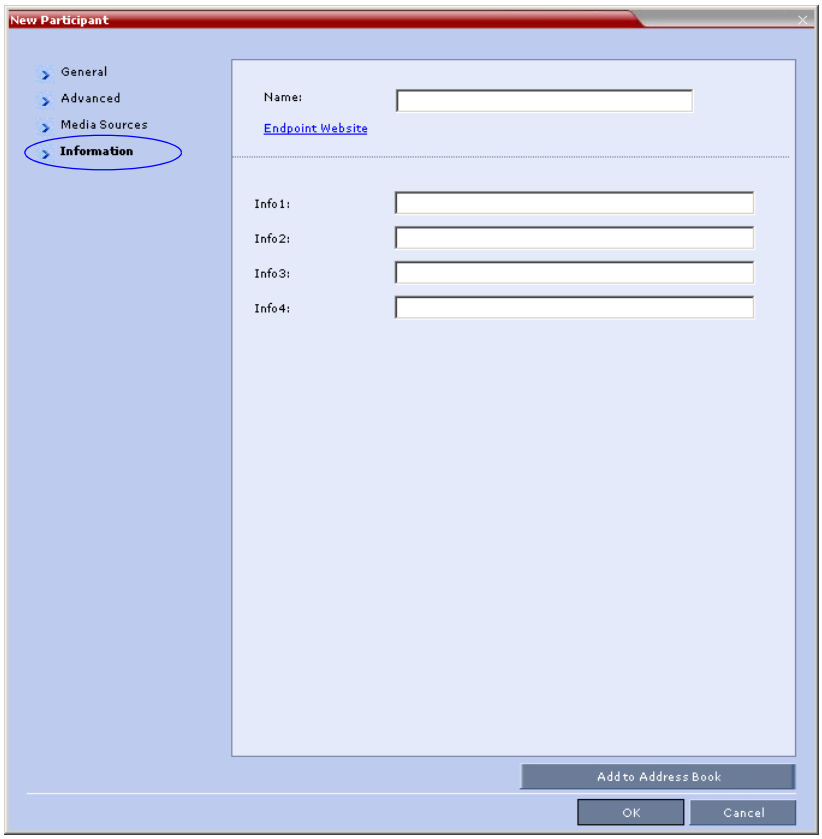
The *Media Sources* tab enables you to set up and save *Personal Layout* and *Video Forcing* settings for each participant. This is especially important when setting up *Telepresence* conferences.

For a full description of *Personal Layout* and *Video Forcing* settings see the *RMX 2000 Getting Started Guide*, "*Changing the Video Layout of a Conference*" on page 3-55 and "*Video Forcing*" on page 3-57.

12 Modify the *Personal Layout* and *Video Forcing* settings for the participant.

13 **Optional.** Click the **Information** tab.

The *New Participant – Information* tab opens.



For a full description of the *Information* fields see the *RMX 2000 Getting Started Guide, "Information Tab"* on page 3-25.

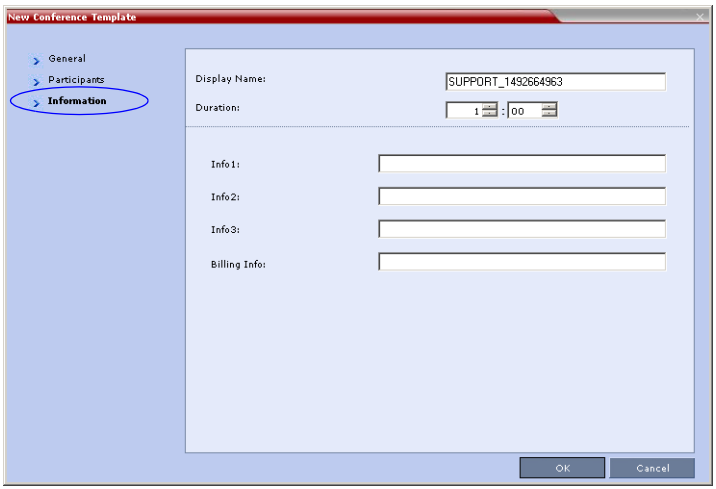
14 Click the **OK** button.

The participant you have defined is added to the *Participants List*.

The New Participant dialog box closes and you are returned to the *New Template – Participant* dialog box (which has remained open since Step 7).

15 **Optional.** In the *New Conference Template* dialog box, click the **Information** tab.

The *New Conference Template – Information* tab opens.



For a full description of the *Information* fields see the *RMX 2000 Getting Started Guide, "Information Tab"* on page **3-25**.

16 Click the **OK** button.

The *New Conference Template* is created and its name is added to the *Conference Templates* list.

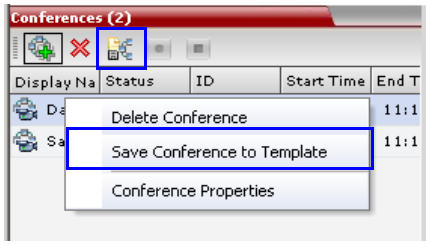
Saving an Ongoing Conference as a Template

Any conference that is ongoing can be saved as a template.

To save an ongoing conference as a template:

1 In the *Conferences List*, select the conference you want to save as a Template.

2 Click the **Save Conference to Template** ( ) button.
or
Right-click and select **Save Conference to Template**.



The conference is saved to a template whose name is taken from the ongoing conference *Display Name*.

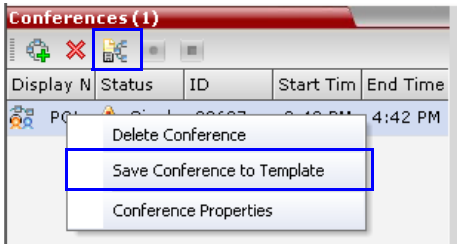
Saving an Operator Conference to a Template

The Operator conference that is ongoing can be saved as a template.

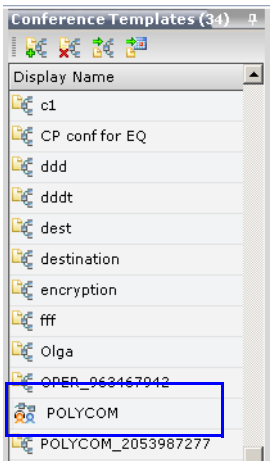
To save an ongoing Operator conference as a template:

1 In the *Conferences List*, select the Operator conference you want to save as a Template.

2 Click the **Save Conference to Template** ( ) button.
or
Right-click and select **Save Conference to Template**.



The conference is saved to a template whose name is taken from the ongoing conference *Display Name* (the Login name of the RMX User). The Template appears with the Operator Conference icon.

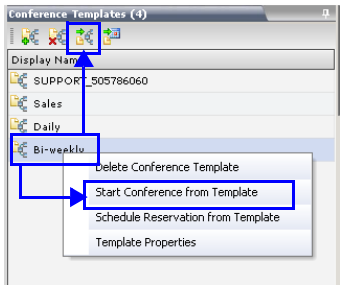


Starting an Ongoing Conference From a Template

An ongoing conference can be started from any Template saved in the *Conference Templates* list.

To start an ongoing conference from a Template:

1. In the *Conference Templates* list, select the Template you want to start as an ongoing conference.
2. Click the **Start Conference from Template** ( ) button.
   or
   Right-click and select **Start Conference from Template**.



The conference is started.

If a Conference Template is assigned a dial-in number that is already assigned to an ongoing conference, Meeting Room, Entry Queue or Gateway Profile, when the template is used to start an ongoing conference or schedule a reservation it will not start. However, the same number can be assigned to several conference templates provided they are not used to start an ongoing conference at the same time. If a dial in number conflict occurs prior to the conference's start time, an alert appears: "ISDN dial-in number is already assigned to another conferencing entity" and the conference cannot start.

The name of the ongoing conference in the *Conferences* list is taken from the Conference Template *Display Name*.

Participants that are connected to other ongoing conferences when the template becomes an ongoing conference are not connected.

If an ongoing conference, Meeting Room or Entry Queue with the same *Display Name, Routing Name* or *ID* already exist in the system, the conference will not be started.

Starting an Operator Conference from a Template

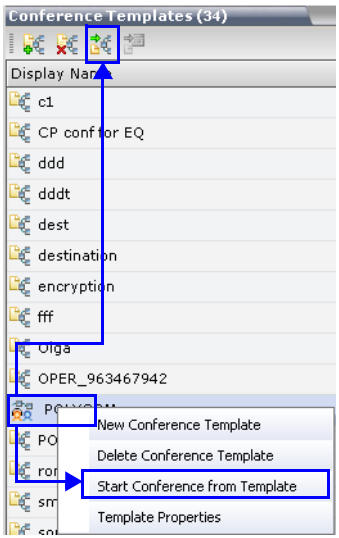
An ongoing Operator conference can be started from an Operator Template saved in the *Conference Templates* list.

To start an ongoing Operator conference from an Operator Template:

1 In the *Conference Templates* list, select the Operator Template to start as an ongoing Operator conference.

- You can only start an Operator conference from a template whose name is identical to your Login Name. For example, if your Login name is Polycom, you can only start an Operator conference from a template whose name is Polycom.
- If an ongoing Operator conference with the same name or any other conference with the same ID is already running, you cannot start another Operator conference with the same login name.

2 Click the **Start Conference from Template** ( ) button.
or
Right-click and select **Start Conference from Template**.



The conference is started.

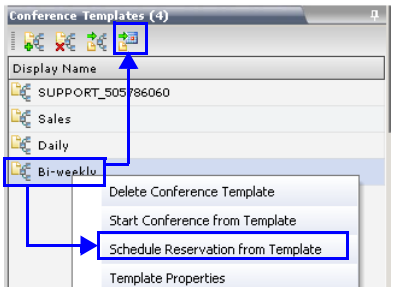
The name of the ongoing conference in the *Conferences* list is taken from the Conference Template *Display Name*.

Scheduling a Reservation From a Conference Template

A *Conference Template* can be used to schedule a single or recurring *Reservation*.

To schedule a Reservation from a Conference Template:

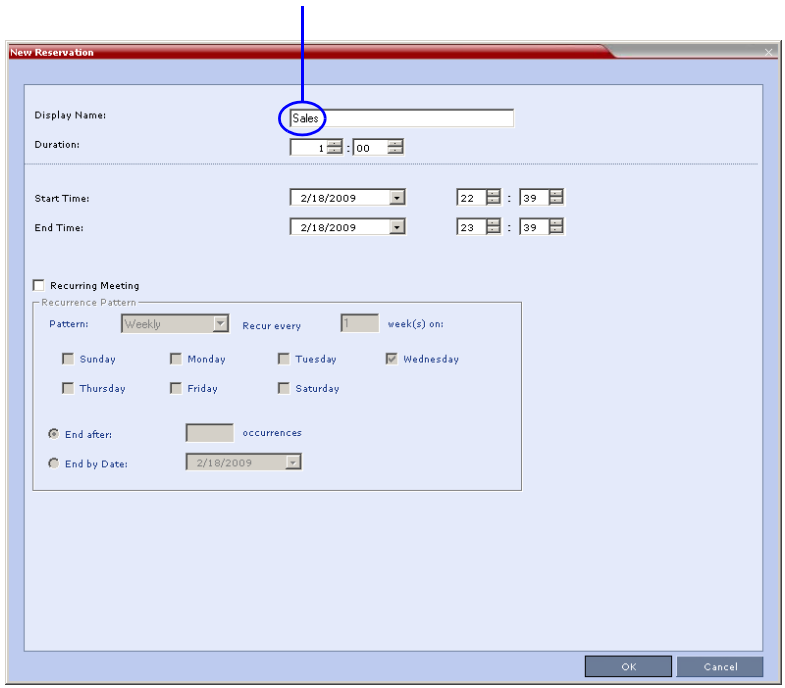
1. In the *Conference Templates* list, select the Conference Template you want to schedule as a Reservation.
2. Click the **Schedule Reservation from Template** ( ) button.
   or
   Right-click and select **Schedule Reservation from Template**.



The *Reservation Properties* dialog box is displayed.

The *Display Name* of the *Reservation* is taken from the Conference Template *Display Name*.

Conference Template and Reservation Name



For a full description of the *Reservation Properties* fields see Table 6-3, "*New Reservation – Schedule Tab,*" on page **6-14**.

3 Modify the fields of the *Reservation Properties*.

4 Click the **OK** button.

A *Reservation* is created based on the *Conference Template*. The *Reservation* can be viewed and modified along with all other *Reservations* using the *Reservations - Calendar View* and *Reservations List*.

If you create a recurring reservation all occurrences have the same ID. A recurring Reservation is assigned the same ISDN/PSTN dial-in number for all recurrences.

If a dial-in number conflict occurs prior to the conference's start time, an alert appears: "ISDN dial-in number is already assigned to another conferencing entity" and the conference cannot start.

The series number (_0000n) of each reservation is appended to its *Display Name*.

Example:

Conference Template name: Sales

Display Name for single scheduled occurrence: Sales

If 3 recurrences of the reservation are created:

Display Name for occurrence 1: Sales_00001

Display Name for occurrence 2: Sales_00002

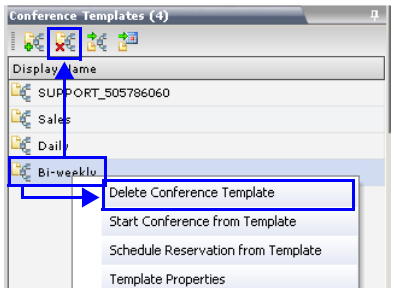
Display Name for occurrence 3: Sales_00003

Deleting a Conference Template

One or several *Conference Templates* can be deleted at a time.

To delete Conference Templates:

1. In the *Conference Templates* list, select the *Template(s)* you want to delete.
2. Click the **Delete Conference Template** ( ) button.
   or
   Right-click and select **Delete Conference Template**.



   A confirmation dialog box is displayed.
3. Click the **OK** button to delete the *Conference Template(s)*.

9

Conference and Participant Monitoring

You can monitor ongoing conferences and perform various operations while conferences are running.

Three levels of monitoring are available with the RMX:

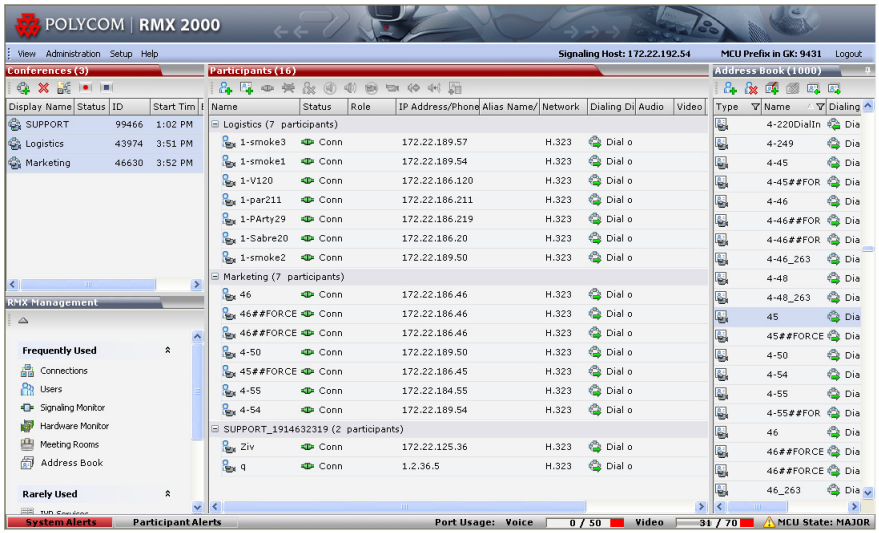
- *General Monitoring* - You can monitor the general status of all ongoing conferences and their participants in the main window.
- *Conference Level Monitoring* - You can view additional information regarding a specific conference and modify its parameters if required, using the *Conference Properties* option.
- *Participant Level Monitoring* - You can view detailed information on the participant's status, using the *Participant Properties* option.

In MPM mode, the maximum number of participants (voice and video) that can connect to a conference is 80.

In MPM+ mode, the maximum number of participants that can connect to a conference is 200. Of these, 80 can be video participants.

General Monitoring

Users can monitor a conference or keep track of its participants and progress. For more information, see *RMX 2000 Getting Started Guide*, "*Monitoring Ongoing Conferences*" on page **3-41**.



You can click the blinking **Participants Alerts** indication bar to view participants that require attention. For more information, see "*System and Participant Alerts*" on page **16-6**.

Conference Level Monitoring

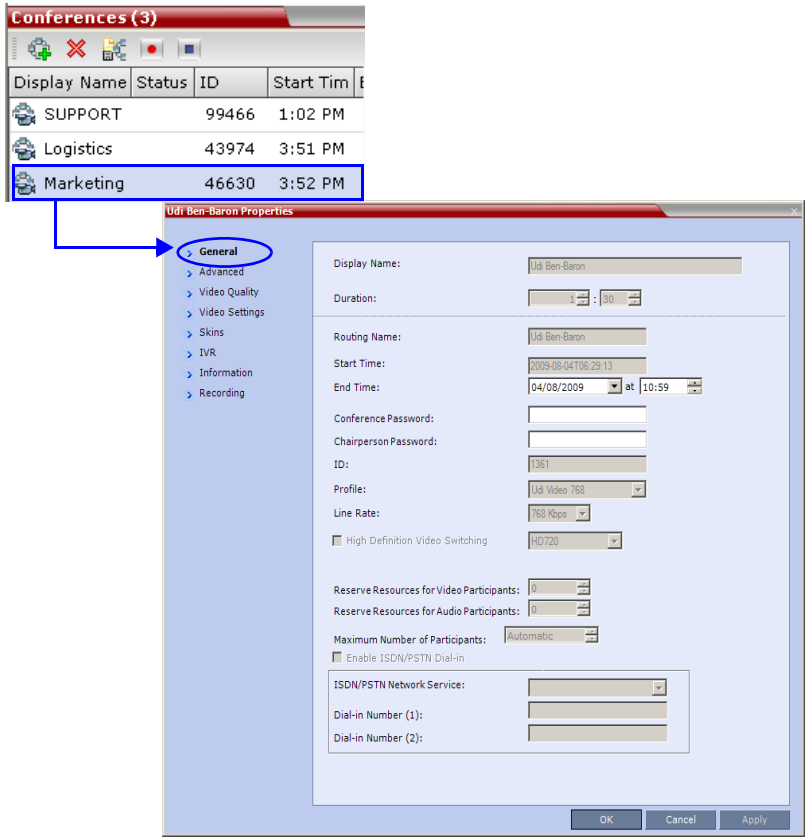
In addition to the general conference information that appears in the *Conference* list pane, you can view the details of the conference's current status and setup parameters, using the *Conference Properties* dialog box.

To view the parameters of an ongoing conference:

1 In the *Conference* list pane, double-click the conference or right-click the conference and then click **Conference Properties**.

The *Conference Properties - General* dialog box with the **General** tab opens.

Viewing Permissions			
Tab	Chairperson	Operator	Administrator
General	✔	✔	✔



The following information appears in the *General* tab:

Table 9-1 *Conference Properties - General*

Field	Description
Display Name	The Display Name is the conference name in native language and Unicode character sets to be displayed in the RMX Web Client. **Note:** This field is displayed in all tabs.
Duration	The expected duration of the conference using the format HH:MM. **Note:** This field is displayed in all tabs.
Routing Name	The ASCII name of the conference. It can be used by H.323 and SIP participants for dialing in directly to the conference. It is used to register the conference in the gatekeeper and the SIP server.
Start Time	The time the conference started.
End Time	The expected conference end time.
Conference Password	A numeric password for participants to access the conference.
Chairperson Password	A numeric password used by participants to identify themselves as the conference chairperson.
ID	The conference ID.
Profile	The name of the conference Profile from which conference parameters were taken.
Line Rate	The maximum transfer rate, in kilobytes per second (Kbps) of the call (video and audio streams).

Table 9-1 *Conference Properties - General (Continued)*

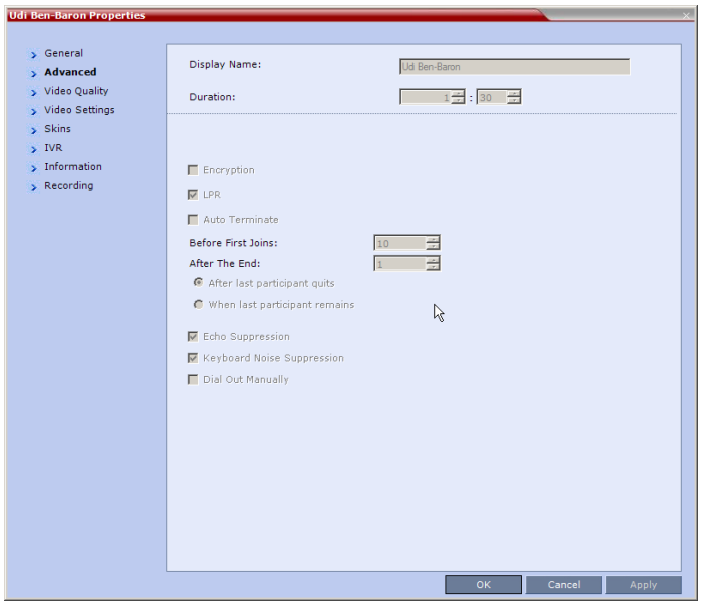
Field	Description
High Definition Video Switching	When selected, the conference is of ultra-high quality video resolution, in a special conferencing mode which implies that all participants must connect at the same line rate and use HD video. This feature utilizes the resources more wisely and efficiently by: • Saving utilization of video ports (1 port per participant as opposed to 4 ports in CP mode). • Video display is in full screen mode only. Drawbacks of this feature are that all participants must connect at the same line rate, (e.g. HD) and all participants with endpoints not supporting HD will connect as secondary (audio only). Video layout changes are not enabled during a conference. High Definition Video Switching supports the following resolutions: • HD 720p • HD 1080p (in MPM+ mode) If HD 1080p is selected, endpoints that do not support HD 1080p resolution are connected as Secondary (Audio Only) participants. **Note:** High Definition Video Switching conferencing mode is unavailable to ISDN participants. For more information, see "*Video Resolutions in CP"* on page **2-3**.
Reserve Resources for Video Participants	Displays the number of video participants for which the system reserved resources. Default: 0 participants.
Reserve Resources for Audio Participants	Displays the number of audio participants for which the system reserved resources. Default: 0 participants.

Table 9-1 *Conference Properties - General (Continued)*

Field	Description
Max Number of Participants	Indicates the total number of participants that can be connected to the conference. The Automatic setting indicates the maximum number of participants that can be connected to the MCU according to resource availability. Irrespective of resource availability, the maximum number of video participants is 80.
Enable ISDN/ PSTN Network Service	When selected, ISDN/PSTN participants can dial into the conference.
ISDN/PSTN Network Service	When the *Enable ISDN/PSTN Network Service* is selected, displays the default Network Service.
Dial-in Number (1)	Displays the conference dial in number.
Dial-in Number (2)	Displays the conference dial in number.

2 Click the **Advanced** tab.

The *Conference Properties - Advanced* dialog box opens.



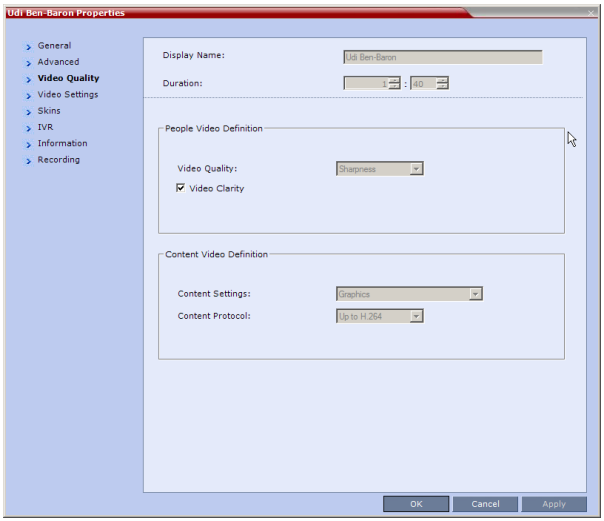
3 The following information appears in the *Advanced* tab:

Table 9-2 *Conference Properties - Advanced Parameters*

Field/Option	Description
Encryption	Indicates whether the conference is encrypted.
LPR	Indicates whether LPR is enabled.
Auto Terminate	When selected, indicates that the MCU will automatically terminate the conference when *Before First Joins*, *At the End-After Last Quits and At the End - When Last Participant Remains* parameters apply.
Echo Suppression	When selected, indicates that when echo is detected it is automatically muted.
Keyboard Noise Suppression	When selected, indicates that when keyboard noises are detected they are automatically muted.
Dial Out Manually	Indicates whether dial-out participants are manually (when selected) or automatically (when cleared) connected to the conference.

4 Click the **Video Quality** tab.

The *Conference Properties - Video Quality* dialog box opens.



The following information appears:

Table 9-3 *Conference Properties - Video Quality Parameters*

Field/Option	Description
People Video Definition	
Video Quality	Indicates the resolution and frame rate that determine the video quality set for the conference. Possible settings are: **Motion** or **Sharpness**. For more information, see "*Video Resolutions in CP"* on page **2-3**.
Video Clarity™	Indicated if Video Clarity is enabled for the conference.
Content Video Definition	
Content Settings	Indicates the Content channel resolution set for the conference. Possible resolutions are: • **Graphics –** default mode • **Hi-res Graphics –** requiring a higher bit rate • **Live Video –** content channel is live video
Content Protocol	**H.263** – Content is shared using *H.263* even if some endpoints have *H.264* capability. **Up to H.264** – *H.264* is the default Content sharing algorithm. When selected: • Content is shared using *H.264* if all endpoints have *H.264* capability. • Content is shared using *H.263* if all endpoints do not have *H.264* capability. • Endpoints that do not have at least *H.263* capability can connect to the conference but cannot share Content.

5 Click the **Video Settings** tab to list the video parameters.

Viewing Permissions			
Tab	Chairperson	Operator	Administrator
Video Settings	✔	✔	✔

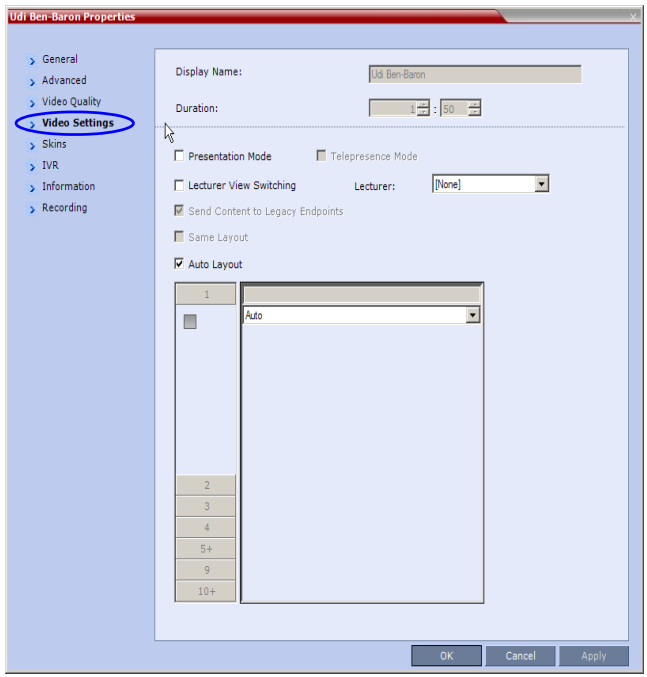


Table 9-4 *Conference Properties - Video Settings Parameters*

Field	Description
Presentation Mode	When checked, indicates that the Presentations Mode is active. For more information, see "*Presentation Mode"* on page **1-16**.
Lecturer View Switching	When checked, the *Lecturer View Switching* enables automatic random switching between the conference participants in the lecturer video window.
Send Content to Legacy Endpoints	Select this option to enable *Legacy* endpoints to send content to H.323/SIP/ISDN endpoints that do not support H.239 Content (legacy endpoints) over the video (people) channel, allowing all conference participants to view the content.

Table 9-4 *Conference Properties - Video Settings Parameters (Continued)*

Field	Description
Same Layout	When checked, forces the selected layout on all conference participants, and the Personal Layout option is disabled.
Auto Layout	When enabled, the system automatically selects the conference layout based on the number of participants in the conference.
Telepresence Mode	Indicates if the conference is running in Telepresence Mode.
Lecturer	Indicates the name of the lecturer (if one is selected). Selecting a lecturer enables the Lecture Mode.
Video Layouts (graphic)	Indicates the currently selected video layout.

Viewing Permissions			
Tab	Chairperson	Operator	Administrator
Skins	✔	✔	✔
IVR		✔	✔
Info	✔	✔	✔

6 Click the **Skins** tab to view the skin selected for the conference.

You cannot select another skin during an ongoing conference.

7 Click the **IVR** tab to view the IVR settings.

8 Click the **Information** tab to view general information defined for the conference. Changes made to this information once the conference is running are not saved to the CDR.

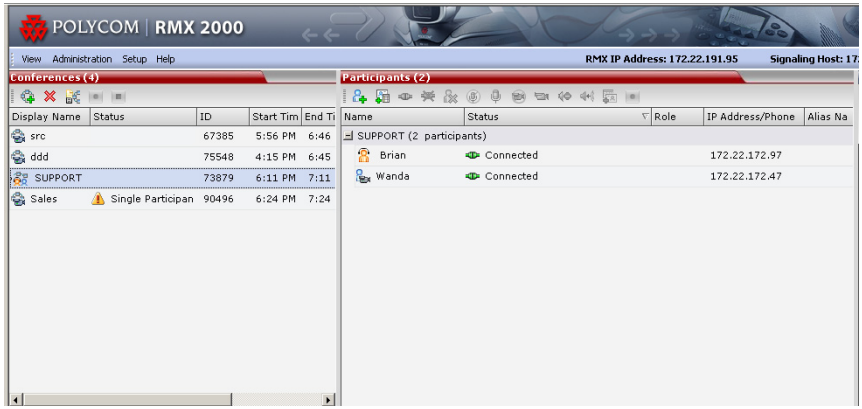
9 Click the **Recording** tab to review the recording settings for the conference.

10 Click **OK** to close the *Conference Properties* dialog box.

Monitoring Operator Conferences and Participants Requiring Assistance

Operator conferences are monitored in the same way as standard ongoing conferences.

Each Operator conference includes at least one participant - the Operator.


POLYCOM | RMX 2000
View Administration Setup Help
RMX IP Address: 172.22.191.95
Signaling Host: 17
Conferences (4)
Display Name Status ID Start Tim End Ti
src 67385 5:56 PM 6:46
ddd 75548 4:15 PM 6:45
SUPPORT 73879 6:11 PM 7:11
Sales Single Participan 90496 6:24 PM 7:24
Participants (2)
Name Status Role IP Address/Phone Alias Na
SUPPORT (2 participants)
Brian Connected 172.22.172.97
Wanda Connected 172.22.172.47

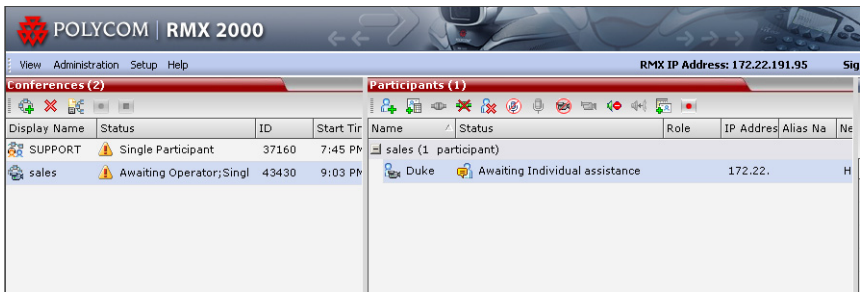
You can view the properties of the *Operator conference* by double-clicking the conference entry in the *Conferences* list or by right-clicking the conference entry and selecting **Conference Properties**. For more information, see the *RMX 2000 Getting Started Guide*, "*Conference Level Monitoring*" on page **3-43**.

Requesting Help

A participant can request help using the appropriate DTMF code from his/her touch tone telephone or the endpoint's DTMF input device. The participant can request *Individual Assistance* (default DTMF code ***0**) or *Conference Assistance* (default DTMF code **00**).

Participants in Entry Queues who failed to enter the correct destination conference ID or the conference password will wait for operator assistance (provided that an Operator conference is active).

When requiring or requesting operator assistance, the RMX management application displays the following:


POLYCOM | RMX 2000
View Administration Setup Help
RMX IP Address: 172.22.191.95
Conferences (2)
Participants (1)
Display Name Status ID Start Tir
SUPPORT Single Participant 37160 7:45 PM
sales Awaiting Operator;Singl 43430 9:03 PM
Name Status Role IP Addres Alias Na
sales (1 participant)
Duke Awaiting Individual assistance 172.22.

- The participant's connection *Status* changes, reflecting the help request. For details, see Table 9-5.
- The conference status changes and it appears with the exclamation point icon and the status "Awaiting Operator".
- The appropriate voice message is played to the relevant participants indicating that assistance will be provided shortly.

The following icons and statuses are displayed in the *Participant Status* column:

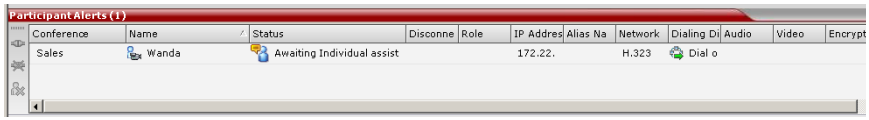
Table 9-5 *Participants List Status Column Icons and Indications*

Icon	Status indication	Description
	Awaiting Individual Assistance	The participant has requested the operator's assistance for himself/ herself.
	Awaiting Conference Assistance	The participant has requested the operator's assistance for the conference. Usually this means that the operator is requested to join the conference.

When the Operator moves the participant to the *Operator conference* for individual assistance the participant Status indications are cleared.

Participant Alerts List

The *Participant Alerts* list contains all the participants who are currently waiting for operator assistance.



Participants are automatically added to the *Participants Alerts* list in the following circumstances:

- The participant fails to connect to the conference by entering the wrong conference ID or conference password and waits for the operator's assistance
- The participant requests Operator's Assistance during the ongoing conference

This list is used as reference only. Participants can be assisted and moved to the *Operator conference* or the destination conference only from the *Participants* list of the Entry Queues or ongoing conference where they are awaiting assistance.

The participants are automatically removed from the *Participant Alerts* list when moved to any conference (including the *Operator conference*).

Participant Level Monitoring

In addition to conference information, you can view detailed information regarding the status and parameters of each listed participant, using the *Participant Properties* dialog box. Participant properties can be displayed for all participants currently connected to a conference and for defined participants that have been disconnected.

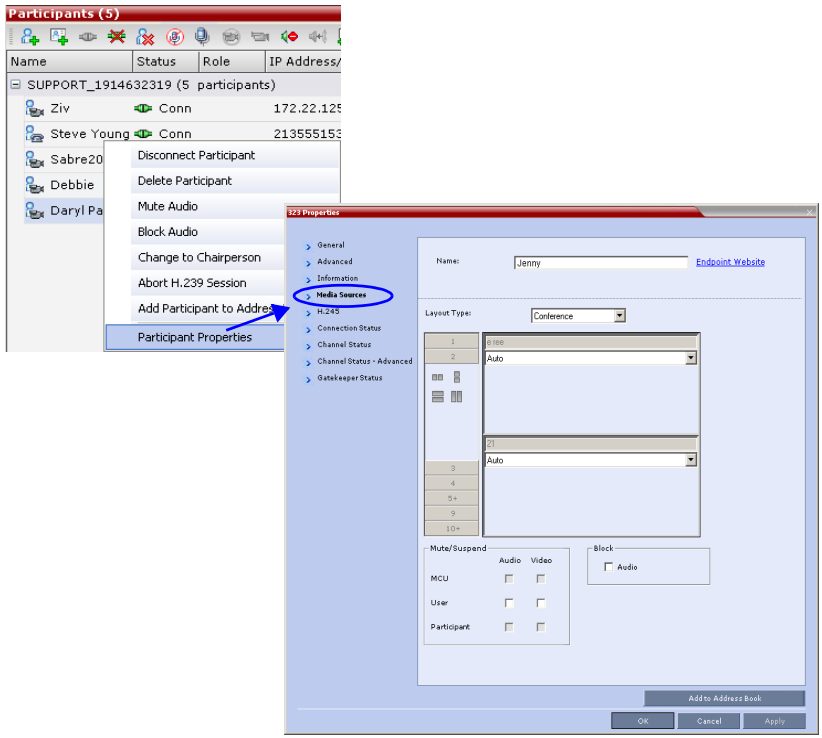
Properties differ for IP and ISDN/PSTN participants.

Displaying Participants Properties:

1 In the *Participant List* pane double-click the participant entry. Alternatively, right-click a participant and then click **Participant Properties**.

The *Participant Properties - Media Sources* dialog box opens.

Viewing Permissions			
Tab	Chairperson	Operator	Administrator
Media Sources	✔	✔	✔



The *Media Sources* dialog box enables you to mute participant's audio, suspend participant's video transmission and select a personal Video Layout for the participant.

For ISDN/PSTN participants, only the following tabs are displayed in the *Participant Properties* dialog box:

- General, Advanced, Information
- Media Sources
- Connection Status
- Channel Status

The *General*, *Advanced* and *Information* tabs include the same properties for new and defined participants. For more information, see "*Adding a new participant to the Address Book Directly"* on page **5-4**.

IP Participant Properties

Table 9-6 *Participant Properties - Media Sources Parameters*

Field	Description
Name	Indicates the participant's name. **Note:** This field is displayed in all tabs.
Endpoint Website	Click the Endpoint Website hyperlink to connect to the internal website of the participant's endpoint. It enables you to perform administrative, configuration and troubleshooting activities on the endpoint. The connection is available only if the IP address of the endpoint's internal site is filled in the *Website IP Address* field in the *Participant Properties - General* dialog box. **Note:** This field is displayed in all tabs (excluding ISDN/PSTN participants).
Layout Type	Indicates whether the video layout currently viewed by the participant is the Conference or Personal Layout. If *Personal Layout* is selected, you can select a Video Layout that will be viewed only by this participant.

Table 9-6 *Participant Properties - Media Sources Parameters (Continued)*

Field	Description
Video Layout	Indicates the video layout currently viewed by the participant. When *Personal Layout* is selected in the *Layout Type* you can force participants to the video windows in a layout that is specific to the participant. For more information*, see RMX 2000 Getting Started Guide,* "*Personal Layout Control with the RMX Web Client"* on page **3-63**.
Mute/Suspend	Indicates if the endpoint's audio and/or video channels from the endpoint have been muted/suspended. The entity that initiated audio mute or video suspend is also indicated. • **MCU** – Audio or Video channel has been muted/ suspended by the MCU. • **User** – Channels have been muted/suspended by the RMX user. • **Participant** – Channels have been muted/ suspended by the participant from the endpoint. You can also cancel or perform mute and suspend operation using these check boxes.
Block	When checked, the audio transmission from the conference to the participant's endpoint is blocked, but the participant will still be heard by other participants.

2 Click the **Connection Status** tab to view the connection status, and if disconnected the cause of the disconnection.

Viewing Permissions			
Tab	Chairperson	Operator	Administrator
Connection Status	✔	✔	✔

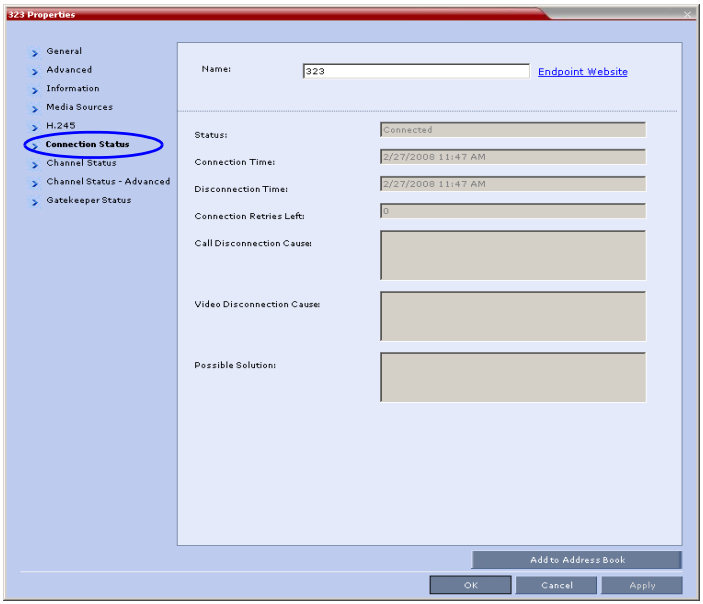


Table 9-7 *Participant Properties - Connection Status Parameters*

Field	Description
Participant Status	
Status	Indicates the connection status of the participant.
Connection Time	The date and time the participant connected to the conference. **Note:** The time format is derived from the MCU's operating system time format.
Disconnection Time	The date and time the defined participant disconnected from the conference.
Connection Retries Left	Indicates the number of retries left for the system to connect defined participant to the conference.

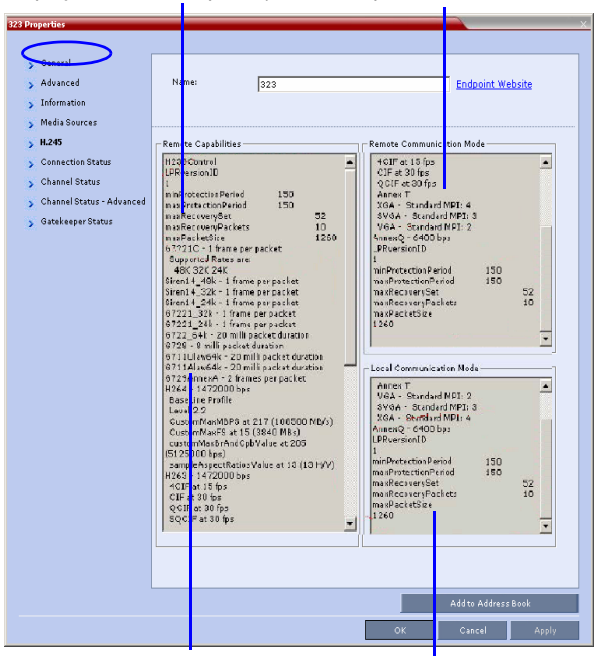
Table 9-7 Participant Properties - Connection Status Parameters (Continued)

Field	Description
Call Disconnection Cause	Displays the cause for the defined participant's disconnection from the conference. See *Appendix A: "Disconnection Causes"* on page **A-1**.
Video Disconnection Cause	Displays the cause the video channel could not be connected. For more information, see *Appendix A: "Disconnection Causes"* on page **A-1**.
Possible Solution	In some cases, a possible solution is indicated to the cause of the video disconnection.

3 Click the **H.245** (H.323) or **SDP** (SIP) tab during or after the participant's connection process to view information that can help in resolving connection issues.

LPR activity (Displayed in all three panes)

Displays the endpoint's actual capabilities used for the connection



List's the endpoint's capabilities as retrieved from the remote site

Displays the MCU's capabilities used for connection with the participant

Table 9-8 *Participant Properties - H.245/SDP Parameters*

Field	Description
Remote Capabilities	Lists the participant's capabilities as declared by the endpoint.
Remote Communication Mode	Displays the actual capabilities used by the endpoint when establishing the connection with the MCU (Endpoint to MCU).
Local Communication Mode	Displays the actual capabilities used by the MCU when establishing the connection with the participant's endpoint (MCU to Endpoint).

4 Click on the **Channel Status** tab to view the status of the various channels.

Viewing Permissions			
Tab	*Chairperson*	*Operator*	*Administrator*
Channel Status		✔	✔

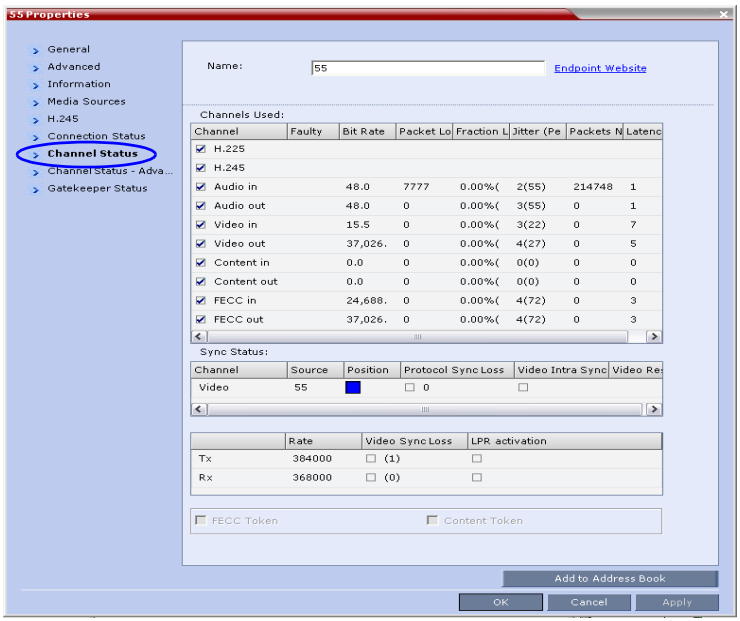


Table 9-9 *Participant Properties - Channel Status Parameters*

Field	Description
Channels Used	When checked, indicates the channel type used by the participant to connect to the conference: Incoming channels are endpoint to MCU, Outgoing channels are from MCU to endpoint. **Channels:** • *H.225/Signaling* - The call-signaling channel. • *H.245/SDP* - The Control channel. • *Audio in - Incoming audio channel* • *Audio out - Outgoing audio channel* • *Video in - Incoming video channel* • *Video out - Outgoing video channel* • *Content in* - H.239/People+Content conferences • *Content out* - H.239/People+Content conferences • *FECC in* - The incoming FECC channel is open. • *FECC out* - The outgoing FECC channel is open. **Columns:** • **Faulty** – A red exclamation point indicates a faulty channel condition. This is a real-time indication; when resolved the indication disappears. An exclamation point indicates that further investigation may be required using additional parameters displayed in the *Advanced Channel Status* tab. • **Bit Rate** – The actual transfer rate for the channel. • **Packet Loss** – The accumulated count of all packets that are missing according to the RTCP report since the channel was opened. This field is relevant only during the connection stage and does not display faulty indications. • **Fraction Loss (Peak)** – The ratio between the number of lost packets and the total number of transmitted packets since the last RTCP report. *Peak* (in parentheses) indicates the highest ratio recorded since the channel was opened.

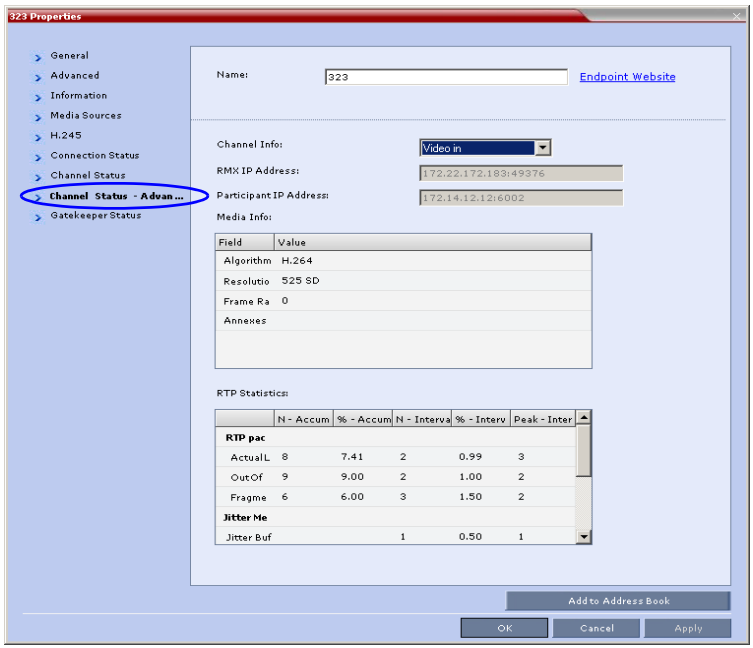
Table 9-9 *Participant Properties - Channel Status Parameters (Continued)*

Field	Description
Channels Used (cont.)	• **Number of Packets** – The number of received or transmitted packets since the channel has opened. This field does not cause the display of the faulty indicator. • **Jitter (Peak)** – Displays the network jitter (the deviation in time between the packets) as reported in the last RTCP report (in milliseconds). *Peak* (in parentheses) reflects the maximum network jitter since the channel was opened. • **Latency** – Indicates the time it takes a packet to travel from one end to another in milliseconds (derived from the RTCP report).
Sync Status	***Channel*** - The channel type: Video or Content. ***Source*** - The name of the participant currently viewed by this participant. ***Position*** - The video layout position indicating the place of each participant as they appear in a conference. ***Protocol Sync Loss*** - Indicates whether the system was able to synchronize the bits order according to the selected video protocol. ***Video Intra Sync*** - Indicates whether the synchronization on a video Intra frame was successful. ***Video Resolution*** - The video resolution of the participant.
Rx - Rate	The received line rate.
Tx - Rate	The transmitted line rate.
Tx - Video Sync Loss	When checked, indicates a video synchronization problem in the outgoing channel from the MCU. The counter indicates the sync-loss count.
Rx - Video Sync Loss	When checked, indicates a video synchronization problem in the incoming channel from the endpoint. The counter indicates the sync-loss count.

Table 9-9 *Participant Properties - Channel Status Parameters (Continued)*

Field	Description
Tx - LPR Activation	When checked, indicates LPR activation in the outgoing channel.
Rx - LPR Activation	When checked, indicates LPR activation in the incoming channel.
FECC Token	When checked, indicates that the participant is the holder of the FECC Token.
Content Token	When checked, indicates that the participant is the holder of the Content Token.

5 Click the **Channel Status Advanced** tab to view additional information for selected audio and video channels.

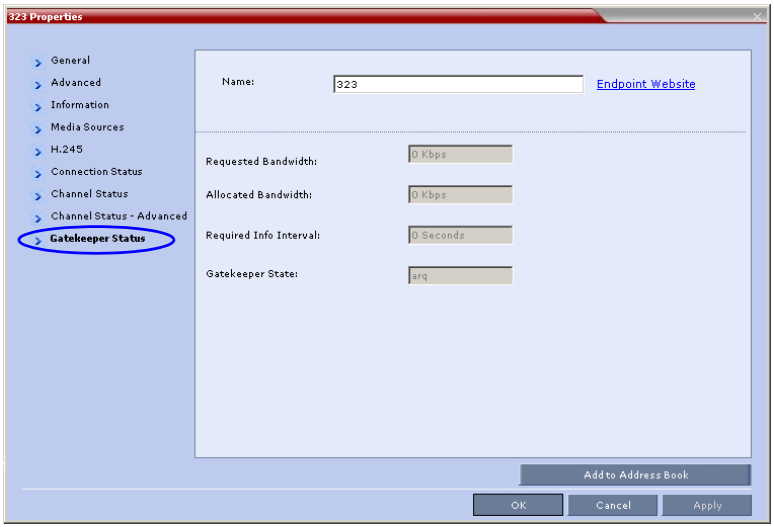


Viewing Permissions			
Tab	Chairperson	Operator	Administrator
Channel Status Advanced			✔

***Table 9-10** Participant Properties - Channel Status Advanced Parameters*

Field	Description
Channel Info	Select a channel to view its information.
MCU Address	The IP address of the MCU to which the participant is connected and the port number allocated to the participant incoming media stream on the MCU side.
Party Address	The IP address of the participant and the port number allocated to the media stream on the participant side.
Media Info	This table provides information about the audio and video parameters, such as video algorithm, resolution, etc.... For more information, see *Appendix E: "Participant Properties Advanced Channel Information"* on page **E-1**.
RTP Statistics	This information may indicate problems with the network which can affect the audio and video quality. For more information, see *Appendix E: "Participant Properties Advanced Channel Information"* on page **E-1**.

6 Click the **Gatekeeper Status** tab to view its parameters.



Viewing Permissions			
Tab	Chairperson	Operator	Administrator
Gatekeeper Status	✔	✔	✔

Table 9-11 *Participant Properties - Gatekeeper Status Parameters*

Field	Description
Requested Bandwidth	The bandwidth requested by the MCU from the gatekeeper.
Allocated Bandwidth	The actual bandwidth allocated by the gatekeeper to the MCU.
Required Info Interval	Indicates the interval, in seconds, between registration messages that the MCU sends to the gatekeeper to indicate that it is still connected.
Gatekeeper State	Indicates the status of the participant's registration with the gatekeeper and the bandwidth allocated to the participant. The following statuses may be displayed: • ***ARQ*** – Admission Request - indicates that the participant has requested the gatekeeper to allocate the required bandwidth on the LAN. • ***Admitted*** – indicates that the gatekeeper has allocated the required bandwidth to the participant. • ***DRQ*** – Disengage Request – the endpoint informs the gatekeeper that the connection to the conference is terminated and requests to disconnect the call and free the resources. • ***None*** – indicates that there is no connection to the gatekeeper.

Monitoring ISDN/PSTN Participants

Using the *Participant Properties* dialog box, you can monitor and verify the properties of an ISDN/PSTN participant. The dialog box's tabs contain information that is relevant to the participant's status only while the conference is running and is used to monitor the participant's status when connection problems occur.

- Table 9-12 lists the audio algorithms that are supported for ISDN participants according to their connection bit rate:

Table 9-12 *Supported Audio Algorithms vs Bit Rate*

	Bit Rate		
	96Kbps (and Lower)	**128Kbps – 192Kbps**	**256Kbps (and Higher)**
Audio Algorithm	G722.1 16K	G722.1 C 32K	G722.1 C 48K
	G722.1 C 24K	G722.1 C 24K	G722.1 C 32K
	Siren14 24K	Siren14 32K	G722.1 C 24K
	G722 48K	Siren14 24K	Siren14 48K
	G722 56K	G722.1 32K	Siren14 32K
	G722 64K	G722.1 24K	Siren14 24K
	G711 56K	G722 48K	G722.1 32K
	G711 64K	G722 56K	G722.1 24K
		G722 64K	G722.1 16K
		G711 56K	G722 48K
		G711 64K	G722 56K
			G722 64K
			G711 56K
			G711 64K

To view the participant's properties during a conference:

1 In the *Participants* list, right click the desired participant and select **Participant Properties**.

The *Participant Properties - Media Sources* dialog box is displayed.

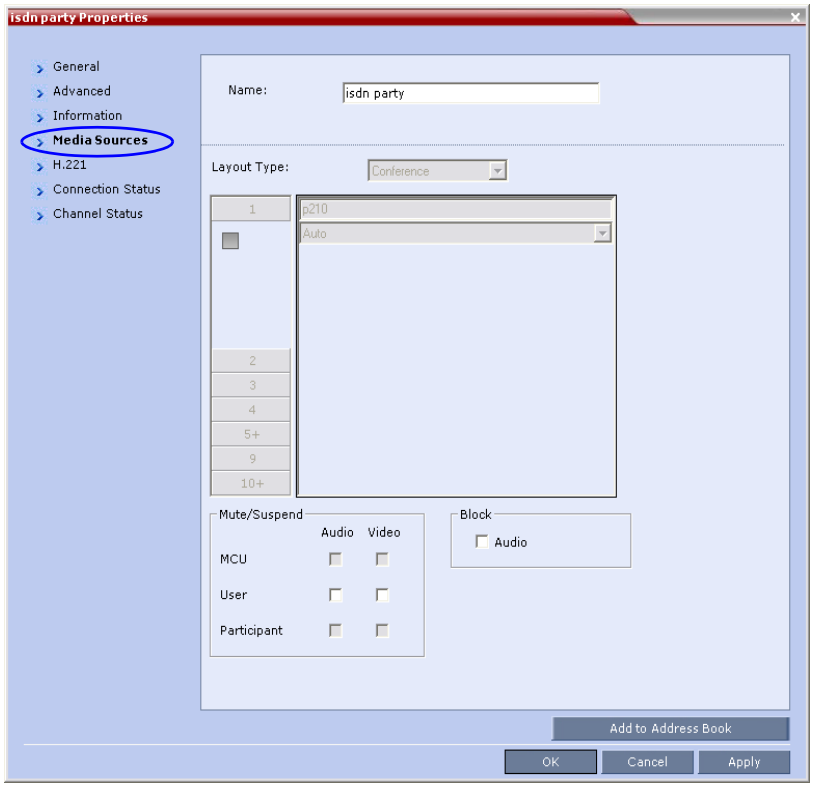


Table 9-13 *ISDN/PSTN Participant Properties - Media Sources*

Field	Description
Mute/Suspend	Indicates if the endpoint's audio and/or video channels from the endpoint have been muted/ suspended.

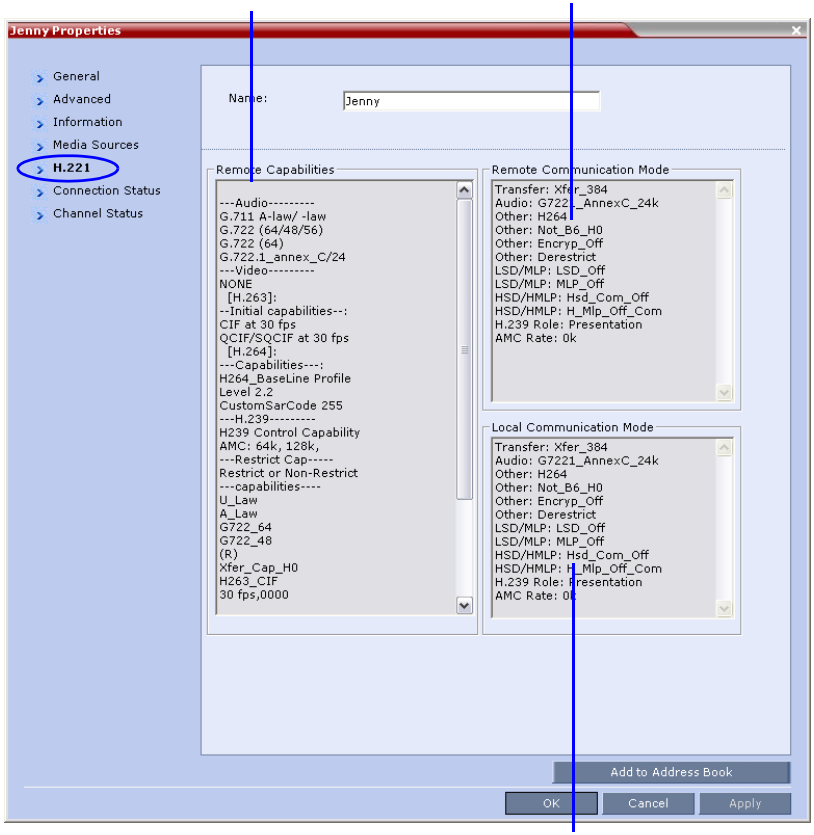
Table 9-13 *ISDN/PSTN Participant Properties - Media Sources (Continued)*

Field	Description
Mute/Suspend (cont.)	The entity that initiated audio mute or video suspend is also indicated. • **MCU** – Audio or Video channel has been muted/suspended by the MCU. • **User** – Channels have been muted/suspended by the RMX user. • **Participant** – Channels have been muted/ suspended by the participant from the endpoint. You can also cancel or perform mute and suspend operation using these check boxes.
Block (Audio)	When checked, the audio transmission from the conference to the participant's endpoint is blocked, but the participant will still be heard by other participants.

2 Click the **H.221** tab to view additional information that can help to resolve connection issues.

List's the endpoint's capabilities as retrieved from the remote site

Displays the endpoint's actual capabilities used for the connection



Displays the MCU's capabilities used for connection with the participant

Table 9-14 *Participant Properties - H.221 Parameters*

Field	Description
Remote Capabilities	Lists the participant's capabilities as declared by the endpoint.

Table 9-14 *Participant Properties - H.221 Parameters (Continued)*

Field	Description
Remote Communication Mode	Displays the actual capabilities used by the endpoint when establishing the connection with the MCU (Endpoint to MCU).
Local Communication Mode	Displays the actual capabilities used by the MCU when establishing the connection with the participant's endpoint (MCU to Endpoint).

3 Click the **Connection Status** tab to view general information regarding the participant connection and disconnection causes of the participant to the conference.

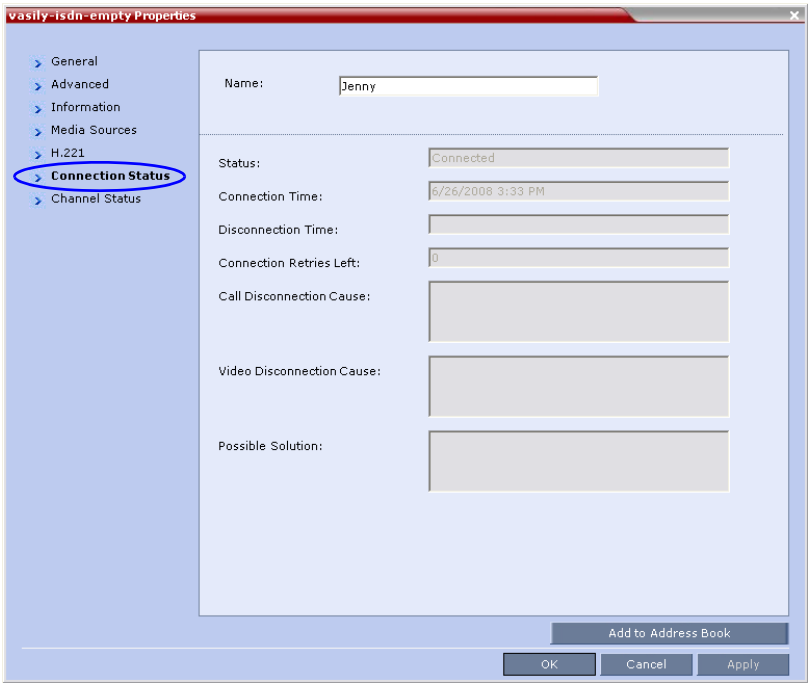


Table 9-15 *ISDN/PSTN Participant Properties - Connection Status*

Field	Description
Status	Indicates the connection status of the participant to the conference. If there is a problem, the appropriate status appears, for example, Disconnected.
Connection Time	The date and time the participant connected to the conference.
Disconnection Time	The date and time the participant was disconnected from the conference.
Connection Retries Left	Indicates the number of retries left for the system to connect the participant to the conference.
Call Disconnection Cause	For a full list of *Disconnection Causes*, see "*ISDN Disconnection Causes"* on page **A-10**.

4 Click the **Channel Status** tab to view the status of a participant's channels.

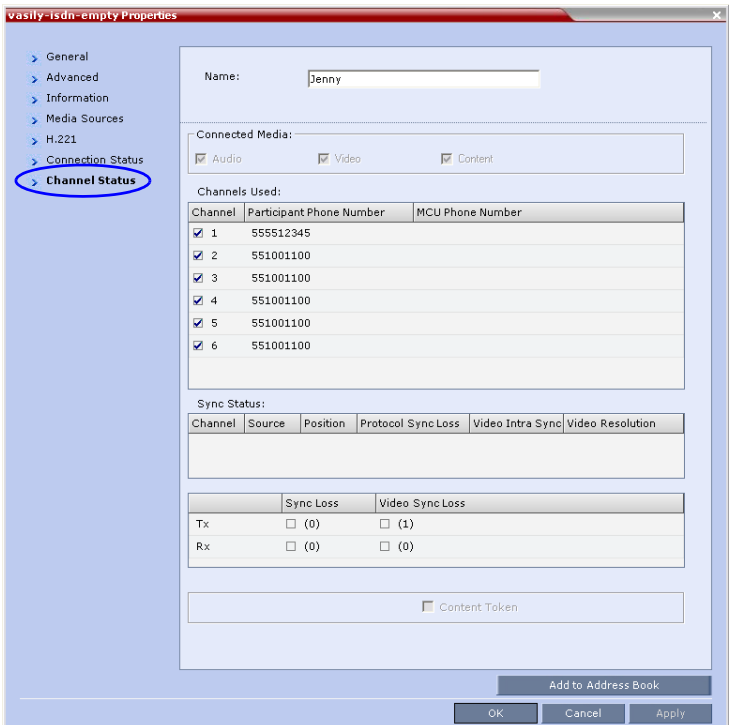


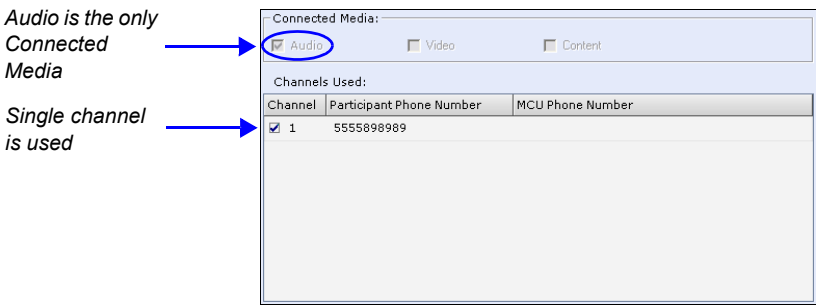
Table 9-16 *ISDN/PSTN Participant Properties - Channel Status*

Field	Description
Connected Media	Indicates if the participant is connected with Audio, Video and Content media channels.
Channels Used	• **Channel** – Indicates the channel used by the participants and whether the channel is connected (indicated with a check mark) or disconnected.

Table 9-16 *ISDN/PSTN Participant Properties - Channel Status*

Field	Description
Channels Used (continued)	• **Participant Phone Number** – In a dial-in connection, indicates the participant's CLI (Calling Line Identification) as identified by the MCU. In a dial-out connection, indicates the participant's phone number dialed by the MCU for each channel. • **MCU Phone Number** – In a dial-in connection, indicates the MCU number dialed by the participant. In a dial-out connection, indicates the MCU (CLI) number as seen by the participant. This is the number entered in the MCU Number field in the Network Service.
Tx - Video Sync Loss	When checked, indicates a video synchronization problem in the outgoing channel from the MCU. The counter indicates the sync-loss count.
Rx - Video Sync Loss	When checked, indicates a video synchronization problem in the incoming channel from the endpoint. The counter indicates the sync-loss count.
Content Token	A check mark indicates that the participant is the current holder of the Content Token.

The *Connected Media* and *Channels Used* fields of an *Audio Only* participant are displayed as follows:



10

Recording Conferences

The RMX enables audio and video recording of conferences using Polycom RSS 2000 recording system. The recording system can be installed at the same site as the conferencing MCU or at a remote site. Several MCU's can share the same recording system.

Recording conferences is enabled via a Recording Link, which is a dial-out connection from the conference to the recording system.

Recording can start automatically, when the first participant connects to a conference, or on request, when the RMX user or conference chairperson initiates it.

Configuring the RMX to enable Recording

To make recording possible, you must set up the following components on the conferencing RMX unit:

- Recording Link – defines the connection between the conference and the recording system.
- Recording-enabled Conference IVR Service – recording DTMF codes and messages must be set in the Conference IVR Service to enable "recording-related" voice messages to be played and to allow the conference chairperson to control the recording process using DTMF codes.
- Recording-enabled Profile – recording must be enabled in the Conference Profile assigned to the recorded conference.

Defining the Recording Link

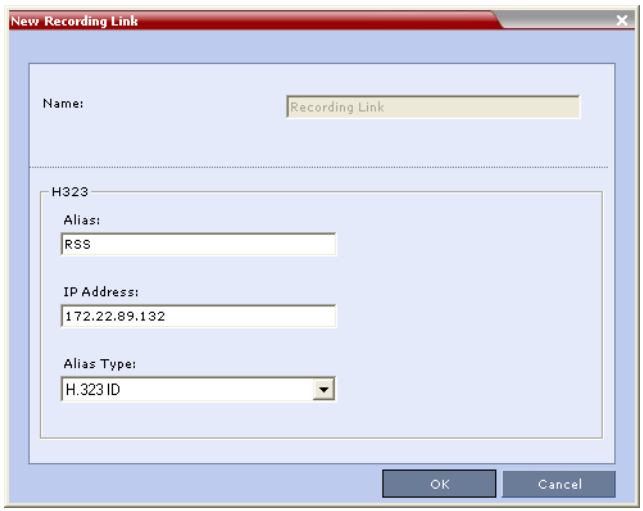
The Recording Link is defined once and can be updated when the H.323 alias or the IP address (of the recording system) is changed. Only one Recording Link can be defined in the RMX and its type must be H.323.

To define a Recording Link:

1 In the *RMX Management* pane, click **Recording Links** ( ).

2 In the *Recording Links* list, click the **New Recording Link** ( ) button.

The *New Recording Link* dialog box is displayed.



3 Define the following parameters:

Table 10-1 *Recording Link Parameters*

Parameter	Description
Name	Displays the default name that is assigned to the Recording Link. This field is disabled and cannot be modified.
IP Address	Enter the IP address of the recording system. Users may either enter the IP Address or Alias or both.
Alias Name / Type	If you are using the endpoint's alias and not the IP address, first select the type of alias and then enter the endpoint's alias: (H.323 ID, E.164, E-mail ID, Participant Number).

4 Click **OK**.

The Recording Link is added to the RMX unit.

Enabling the Recording Features in a Conference IVR Service

In order to record a conference, a Conference IVR Service in which the recording messages and DTMF codes are activated must be assigned to the conference. The default Conference IVR Service shipped with the RMX includes the recording-related voice messages and default DTMF codes that enable the conference chairperson to control the recording process from the endpoint. You can modify these default settings.

To modify the default recording settings for an existing Conference IVR Service:

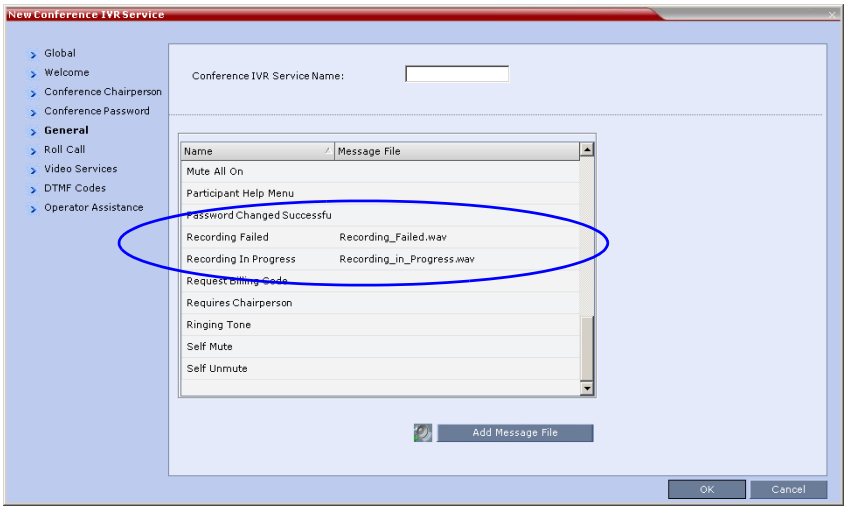
1. In the *RMX Management* pane, click the **IVR Services** ( ) button.

   The IVR Services are listed in the *IVR Services* list pane.

2. To modify the default recording settings, double-click the Conference IVR Service or right-click and select **Properties**.

   The *Conference IVR Service Properties* dialog box is displayed.

3. To assign voice messages other than the default, click the **General** tab and scroll down the list of messages to the recording messages.



4. Select the *Recording In Progress* message, and then select the appropriate message file (by default, Recording_in_Progress.wav) from the file list to the right of the field.

5 Select the *Recording Failed* message, and then select the appropriate message file (by default, Recording_Failed.wav) from the file list to the right of the field.

6 To modify the default DTMF codes, click the **DTMF Codes** tab.

7 To modify the DTMF code or permission for a recording function:

a Select the desired DTMF name (Start, Stop or Pause Recording), click the DTMF code entry and type a new code.

Table 10-2 Default DTMF Codes assigned to the recording process

Recording Operation	DTMF Code	Permission
Start or Resume Recording	***73**	Chairperson
Stop Recording	***74**	Chairperson
Pause Recording	***75**	Chairperson

b In the *Permission* entry, select whether this function can be used by all conference participants or only the chairperson.

8 Click **OK**.

Enabling the Recording in the Conference Profile

To be able to record a conference, the recording options must be enabled in the Conference Profile assigned to it. You can add recording to existing Profiles by modifying them.

To enable recording for a conference:

1 In the *RMX Management* pane, click the **Conference Profiles** ( ) button.

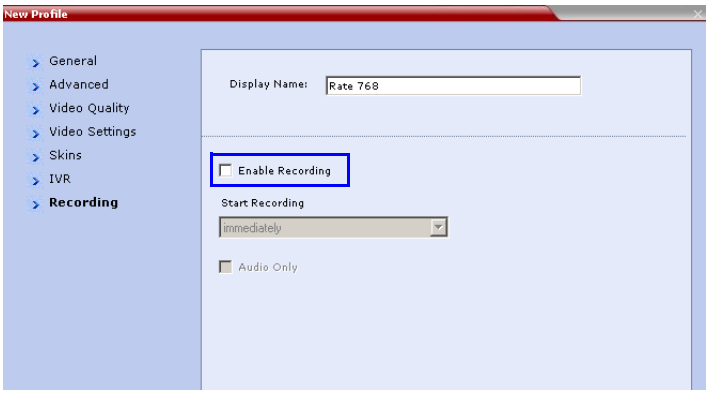
The *Conference Profiles* list is displayed.

2 Create a new profile by clicking the **New Profile** ( ) button or modify an existing profile by double-clicking or right-clicking an existing profile and then selecting **Profile Properties**.

If creating a new profile, complete the conference definition. For more information on creating Profiles see the *RMX Administrators Guide*, *Defining Profiles* on page 1-8.

3 In the *Profile Properties* dialog box, click the **Recording** tab.

4 Select the **Enable Recording** check box.



5 Define the following parameters:

Table 10-3 *Conference Profile Recording Parameters*

Parameter	Description
Start recording	Select one of the following: • **Immediately** – conference recording is automatically started upon connection of the first participant. • **Upon Request** – the operator or chairperson must initiate the recording (manual).
Audio only	Select this option to record only the audio channel of the conference.

6 Click **OK**.
Recording has been enabled for the Conference Profile.

Managing the Recording Process

When a conference is started and recording is enabled in its Profile, the system will automatically start the recording if the *Start Recording* parameter is set to *immediately*. If it is set to *Upon Request*, the system waits for the chairperson or RMX user's request. Once the recording is initiated for a conference, the MCU connects to the Recording device (RSS 2000) using the default Recording Link. The connection that is created between the conference and the recording device is represented as a special participant (Recording) whose name is the Recording Link. Once the recording has started, the recording process can be stopped and restarted from the Chairperson's endpoint (using DTMF codes) or from the RMX Web Client. After the recording process has finished, the recording can be identified in the RSS 2000 by its RMX conference name.

A conference participant and the Recording Link cannot have identical names, otherwise the recording process will fail.

Using the RMX Web Client to Manage the Recording Process

To manage the recording process using the right-click menu:

- Right-click the *Recording* participant in the conference and select from one of the following options:

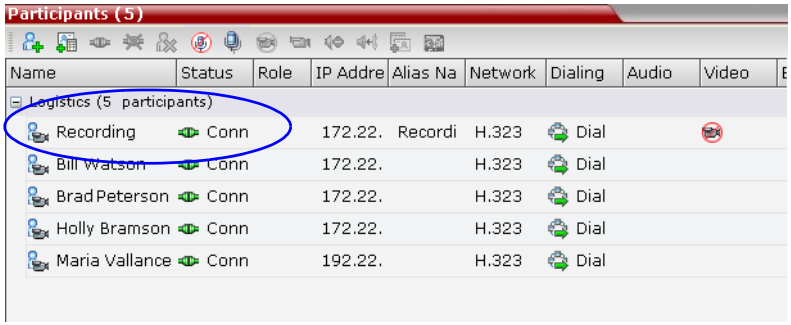
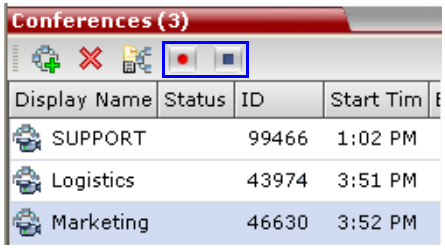


Table 10-4 *Recording Participant Right-click Options*

Name	Description
Start	Starts recording. When recording has started, this option toggles with the *Pause* option.
Pause	Pauses the recording of the conference without disconnecting. When the Recording is Paused, this option toggles with the *Start* option.
Resume	Resumes the recording of the conference. The Resume option toggles with the *Pause* option when it is used.
Stop	Stops the recording. **Note:** The Stop button is only enabled when the Recording is *Started* or *Paused*.
Suspend Video	The Suspend Video option prevents the incoming video of the recording link participant to be part of the conference layout. The Recording Link participant is set by default to Suspend Video. The Suspend Video option toggles with the Resume Video option.
Resume Video	The Resume Video option enables the incoming video of the recording link participant to be part of the conference layout. This feature may be used to play back previously recorded video or audio feeds in the conference layout. For more information, see the RSS 2000 User Guide.
Participant Properties	The Participant Properties option displays viewing only information for monitoring, e.g. communication capabilities and channels used to connect to the conference. Users will not be able to perform any functional requests from this window, i.e. disconnect, change layout and mute.

To manage the recording process using the Conference toolbar:

- In the *Conferences* pane, click one of the following buttons in the Conference toolbar.



The recording buttons will only be displayed in the conference toolbar for a conference that is recording-enabled.

Table 10-5 *Conferences List - Recording Toolbar buttons*

Button	Description
	Start/Resume recording. This button toggles with the *Pause* button.
	Stop recording.
	Pause recording. This button toggles with the *Start/Resume* button.

Using DTMF Codes to Manage the Recording Process

By entering the appropriate DTMF code on the endpoint, the chairperson can **Stop** the recording (*74), **Pause** it (*75), or **Start/Resume** the recording (*73). For more information on managing the recording process via DTMF codes, see the *RSS 2000 User's Guide*.

Conference Recording with Codian IP VCR

Conference recording is available with Codian VCR 2210, VCR 2220 and VCR 2240.

Recording between the RMX and the Codian VCR is enabled by adding an IP participant to the recorded conference that acts as a link between the conference and the recording device. This participant is identified as a recording link to the Codian VCR according to the product ID sent from the VCR during the connection phase, in the call setup parameters.

The video channel between the conference and the recording device is unidirectional where the video stream is sent from the conference to the recorder.

If the Codian VCR opens a video channel to the conference - this channel is excluded from the conference video mix.

To record a conference running on the RMX using Codian recorder:

- In the conference, define or add a dial-out participant using the Codian VCR IP address as the address for dialing.

  Once added to the conference, the MCU automatically connects the participant (the link to Codian VCR) and the recording is automatically started on the Codian VCR.

A connection can also be defined on the Codian VCR, dialing into the recorded conference using the MCU prefix and the conference ID as for any other dial-in participant in the conference.

Monitoring the recording participant:

This connection is monitored as any other participant in the conference. The connection can also be monitored in the Codian VCR web client.

11

Users, Connections and Notes

RMX Web Clients users are defined in the User's table and can connect to the MCU to perform various operations.

The RMX supports four user authorization levels:

- Chairperson
- Operator
- Administrator
- Auditor

The authorization level dictates a user's capabilities within the system.

A **Chairperson** can only manage ongoing conferences and participants. The Chairperson does not have access to the RMX configurations and utilities.

An Operator can perform all the RMX tasks a Chairperson does. In addition, Operators can manage Meeting Rooms, Profiles, Entry Queues, and SIP Factories, and can also view the RMX configurations, but cannot change them.

An Administrator can perform all the tasks of Chairpersons and Operator users. In addition, Administrators can define and delete other users, and perform all configuration and maintenance tasks.

An **Auditor** can only view *Auditor Files* and audit the system.

Administrator and Operator users can verify which users are defined in the system. Neither of them can view the user passwords, but an Administrator can change a password.

The *Users* pane lists the currently defined users in the system and their authorization levels. The pane also enables the administrators to add and delete users.

The RMX is shipped with a default Administrator user called POLYCOM, whose password is POLYCOM. However, once you have defined other authorized Administrator users, it is recommended to remove the default user.

A maximum of 100 users can be defined per RMX.

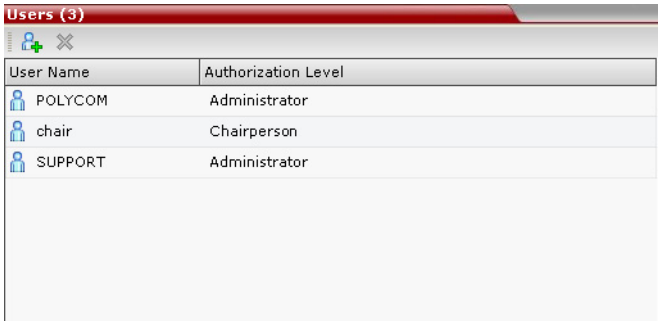
Listing Users

You can view the list of users that are currently defined in the system.

To view the users currently defined in the system:

1 In the *RMX Management* pane, click the **Users** ( ) button.

The *Users* pane appears.



The list includes two columns: User Name and Authorization Level. The User name is the login name used by the user to connect to the RMX.

Adding a New User

Administrators can add new users to the system.

The User Name and Password must be in ASCII.

To add a new user to the system:

1. In the *RMX Management* pane, click the **Users** ( ) button.
2. The *Users* pane appears.
3. Click the **New User** ( ) button or right-click anywhere in the pane and then click **New User**.

   The *New User Properties* dialog box opens.

4. In the *User Name* text box, enter the name of the new user. This is the login name used by the user when logging into the system.
5. In the *Password* text box, enter the new user's password. This will be the user's password when logging into the system.
6. In the *Authorization Level* list, select the user type: **Administrator, Operator, Chairperson** or **Auditor**.
7. Click **OK**.

   The *User Properties* dialog box closes and the new user is added to the system.

Deleting a User

To delete a user, you must have Administrator authorization. The last remaining Administrator in the *Users* list cannot be deleted.

1 In the *RMX Management* pane, click the **Users** ( ) button.

2 Select the user and click the **Delete** ( ) button or right-click the user and then click **Delete User**.

The system displays a confirmation message.

3 In the *confirmation* dialog box, select **Yes** to confirm or **No** to cancel the operation.

If you select **Yes**, the user name and icon are removed from the system.

Changing a User's Password

Users with Administrator authorization can change their own password and other users' passwords. Users with Operator authorization can change their own password.

To change a user's password:

1 In the *RMX Management* pane, click the **Users** ( ) option.

2 Right-click the user and click **Change User Password**.

The *Change Password* dialog box opens.

3 Enter the *Old Password* (current), *New Password* and *Confirm the New Password*.

The Password must be in ASCII.

4 Click **OK**.

The user's password is changed.

Connections

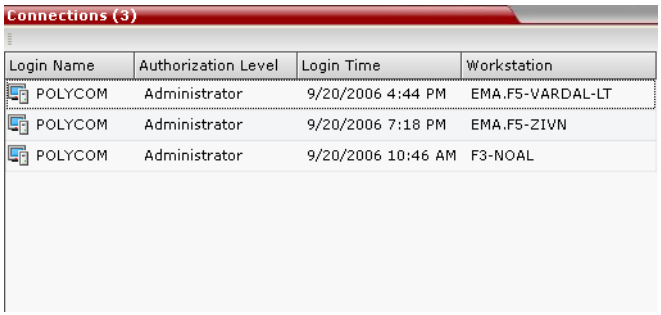
The RMX enables you to list all connections that are currently logged into the MCU, e.g. users, servers or API users. The MCU issues an ID number for each login. The ID numbers are reset whenever the MCU is reset.

A maximum of 50 users can be concurrently logged in to the RMX.

Viewing the Connections List

To list the users who are currently connected to the MCU:

1 In the *RMX Management* pane, click the **Connections** ( ) button.
A list of connected users appears in the *Connections* pane.



The information includes:

— The user's login name.
— The user's authorization level (Chairperson, Operator, Administrator or Auditor).
— The time the user logged in.
— The name/identification of the computer used for the user's connection.

Notes

Notes are the electronic equivalent of paper sticky notes. You can use notes to write down questions, important phone numbers, names of contact persons, ideas, reminders, and anything you would write on note paper. *Notes* can be left open on the screen while you work.

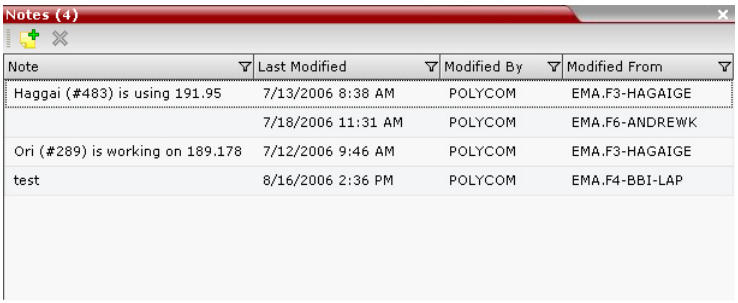
Notes can be read by all RMX Users concurrently connected to the MCU. Notes that are added to the *Notes* list are updated on all workstations by closing and re-opening the *Notes* window. Notes can be written in any Unicode language.

Using Notes

To create a note:

1 On the *RMX* menu, click **Administration > Notes**.

The *Notes* window opens.



2 In the *Notes* toolbar, click the **New Note** ( ) button, or right-click anywhere inside the *Notes* window and select **New Note**.

3 In the *Note* dialog box, type the required text and click **OK**.

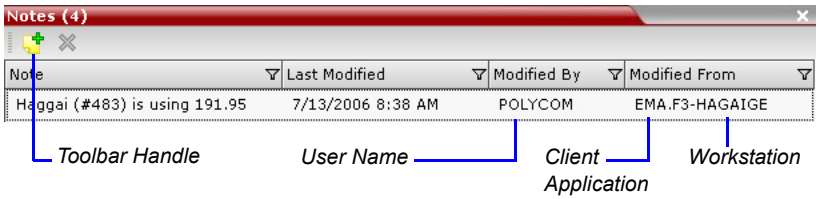
The new note is saved and closed. The *Notes* list is updated, listing the new note and its properties:

— **Note –** The beginning of the note's text.

— **Last Modified –** The date of creation or last modification.

— **Modified By –** The *Login Name* of the user who last modified the note.

— **Modified From –** The *Client Application* and *Workstation* from which the note was created or modified.



To open or edit a note:

- Double-click the entry to edit, or right-click the entry and select **Note Properties**.

  The note opens for viewing or editing.

To delete a note:

1. In the *Notes* list, select the entry for the note to delete and click the **Delete Note** button ( ), or right-click the entry and select **Delete Note**.

   A *delete confirmation* dialog box is displayed.

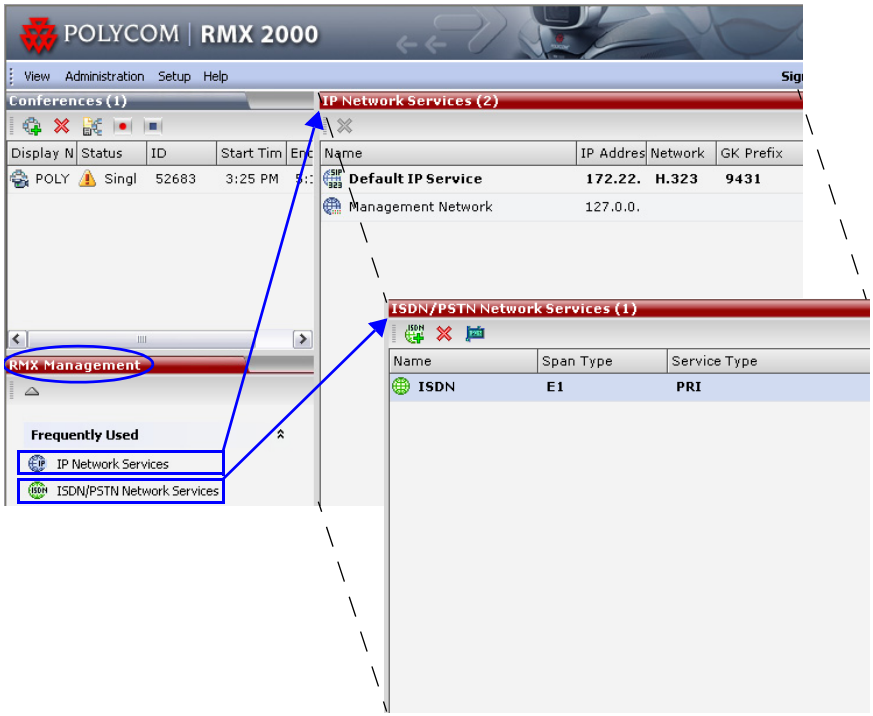
2. Click **OK** to delete the note, or click **Cancel** to keep the note.

12

Network Services

To enable the RMX to function within IP and ISDN/PSTN network environments, network parameters must be defined for both the *IP Network Services* and *ISDN/PSTN Network Services*. The IP Network Service must be defined for the RMX, while the ISDN/PSTN Network Service definition is optional and is done when the RTM ISDN cards are installed in the MCU.

The configuration dialog boxes for both these network services are accessed via the *RMX Management* pane of the *RMX Web Client*.



IP Network Services

Two *IP Services* are defined for the RMX:

- **Management Network**
- **Default IP Service (Conferencing Service)**

Management Network (Primary)

The *Management Network* is used to control the RMX, mainly via the *RMX Web Client* application. The *Management Network* contains the network parameters that connect between the RMX and the *RMX Web Client*.

Management Network parameters include the IP address of the *Control Unit*. This IP address can be used by the administrator or service personnel to connect to the *Control Unit* should the RMX become corrupted or inaccessible. For more information, see *Appendix G, "Alternate Management Network"* on page **G-1**.

During *First Time Power-up,* the *Management Network* parameters can be set either via a *USB key* or by using a cable to create a private network. For more information, see the *RMX 2000 Getting Started Guide, "Procedure 3: First-time Power-up and Connection to MCU"* on page **2-9**.

Default IP Service (Conferencing Service)

The *Default IP Service (Conferencing Service)* is used to configure and manage communications between the RMX and conferencing devices such as endpoints, gatekeepers, SIP servers, etc.

The *Default IP Service* contains parameters for:

- Signaling Host IP Address
- MPM boards (media processors)
- External conferencing devices

Calls from all external IP entities are made to the *Signaling Host,* which initiates call set-up and assigns the call to the appropriate *MPM board*.

Conferencing related definitions such as environment (H.323 or SIP) are also defined in this service.

Most of the *Default IP Service* is configured by the *Fast Configuration Wizard*, which runs automatically should the following occur:

- First time power-up
- *Default IP Service* is deleted, followed by an RMX reset.

For more information, see the *RMX 2000 Getting Started Guide*, "*Procedure 3: First-time Power-up and Connection to MCU*" on page **2-9**.

Changes made to any of these parameters only take effect when the RMX unit is reset. An *Active Alarm* is created when changes made to the system have not yet been implemented and the MCU must be reset.

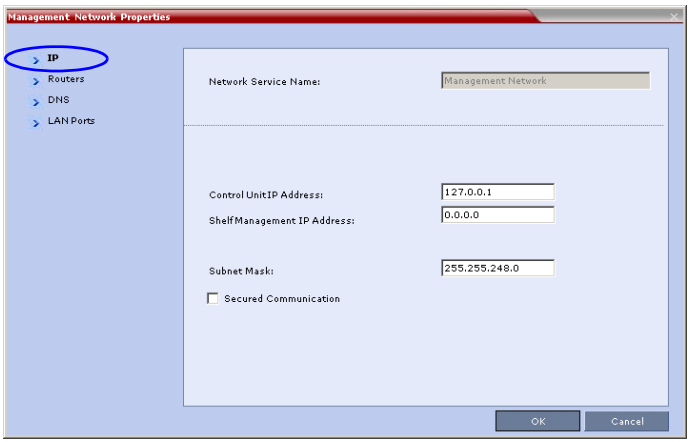
Modifying the Management Network

The *Management Network* parameters need to be modified if you want to:

- Connect directly to the MCU from a workstation
- Modify routes
- Modify DNS information
- Modify the speed of LAN 2 port

To view or modify the Management Network Service:

1. In the *RMX Management* pane, click the **IP Network Services** ( ) button.
2. In the *IP Network Services* list pane, double-click the **Management Network** ( ) entry.
   The *Management Network Properties - IP* dialog box opens.

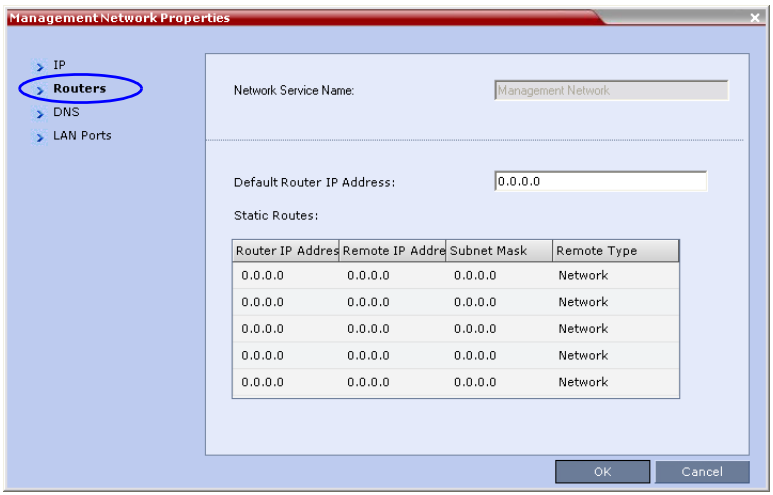


3 View or modify the following fields:

Table 12-1 *Management Network Properties – IP Parameters*

Field	Description
Network Service Name	Displays the name of the Management Network. This name cannot be modified. **Note:** The field is displayed in all Management Network Properties tabs.
Control Unit IP Address	The IP address of the RMX Control Unit. This IP address is used by the RMX Web Client to connect to the RMX.
Shelf Management IP Address	The IP Address of the server that manages the RMX hardware shelf.
Subnet Mask	The subnet mask of the Control Unit.
Secured Communication	Select to enable Secured Communication. The RMX supports TLS 1.0 and SSL 3.0 (Secure Socket Layer). A SSL/TLS Certificate must installed on the RMX for this feature to be enabled. For more information see "*Secure Communication Mode"* on page **F-1**.

4 Click the **Routers** tab.



5 View or modify the following fields:

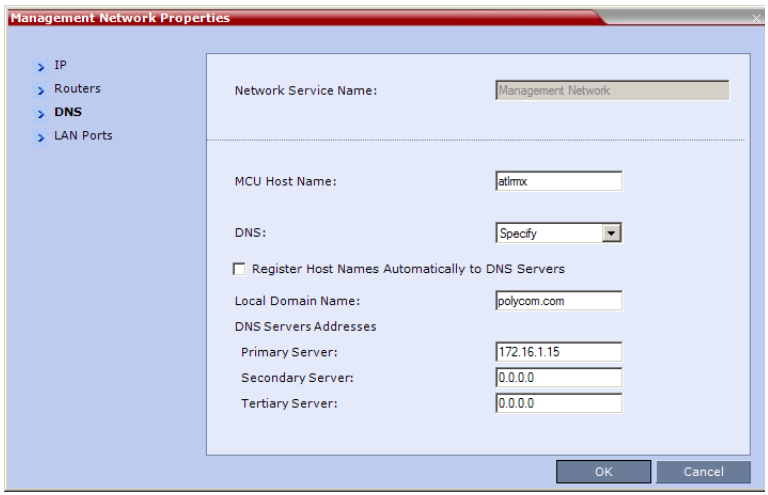
Table 12-2 *Management Network Properties – Routes Parameters*

Field	Description
Default Router IP Address	Enter the IP address of the default router. The default router is used whenever the defined static routers are not able to route packets to their destination. The default router is also used when host access is restricted to one default router.

Table 12-2 *Management Network Properties – Routes Parameters (Continued)*

Field	Description	
Static Routes Table	The system uses *Static Routes* to search other networks for endpoint addresses that are not found on the local LAN. Up to five routers can be defined in addition to the Default Router. The order in which the routers appear in the list determines the order in which the system looks for the endpoints on the various networks. If the address is in the local subnet, no router is used. To define a static route (starting with the first), click the appropriate column and enter the required value.	
	Router IP Address	Enter the IP address of the router.
	Remote IP Address	Enter the IP address of the entity to be reached outside the local network. The *Remote Type* determines whether this entity is a specific component (Host) or a network. • If Host is selected in the *Remote Type* field, enter the IP address of the endpoint. • If Network is selected in the *Remote Type* field, enter of the segment of the other network.
	Remote Subnet Mask	Enter the subnet mask of the remote network.
	Remote Type	Select the type of router connection: • **Network** – defines a connection to a router segment in another network. • **Host** – defines a direct connection to an endpoint found on another network.

6 Click the **DNS** tab.



7 In the *DNS* field, select **Specify** to define the DNS parameters.

8 View or modify the following fields:

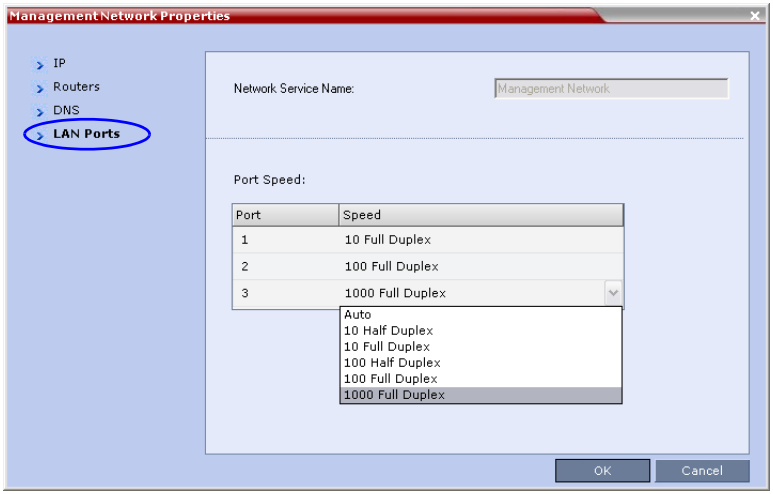
Table 12-3 *Management Network Properties – DNS Parameters*

Field	Description
MCU Host Name	Enter the name of the MCU on the network. Default name is RMX
DNS	Select: • **Off** – if DNS servers are not used in the network. • **Specify** – to enter the IP addresses of the DNS servers. **Note:** The IP address fields are enabled only if **Specify** is selected.
Register Host Names Automatically to DNS Servers	Select this option to automatically register the MCU Signaling Host and Shelf Management with the DNS server.
Local Domain Name	Enter the name of the domain where the MCU is installed.

Table 12-3 *Management Network Properties – DNS Parameters (Continued)*

Field	Description
DNS Servers Addresses:	
Primary Server	The static IP addresses of the DNS servers. A maximum of three servers can be defined.
Secondary Server	
Tertiary Server	

9 Click the **LAN Ports** tab.



10 View or modify the following fields:

Table 12-4 Management Network Properties – LAN Ports Parameters

Field	Description	
Port Speed	The RMX has 3 LAN ports. The administrator can set the speed and transmit/receive mode manually for **LAN 2** Port only.	
	Port	The LAN port number: 1, 2 or 3. **Note:** Do not change the automatic setting of Port 1 and Port 3. Any change to Port 1 speed will not be applied.
	Speed	Select the speed and transmit/receive mode for each port. Default: Auto – Negotiation of speed and transmit/receive mode starts at 1000 Mbits/second Full Duplex, proceeding downward to 10 Mbits/second Half Duplex. **Note:** To maximize conferencing performance, especially in high bit rate call environments, a 1Gb connection is recommended.

11 Click **OK.**

12 If you have modified the *Management Network Properties*, reset the MCU.

Modifying the Default IP Network Service

The *Default IP Service* parameters need to be modified if you want to change the:

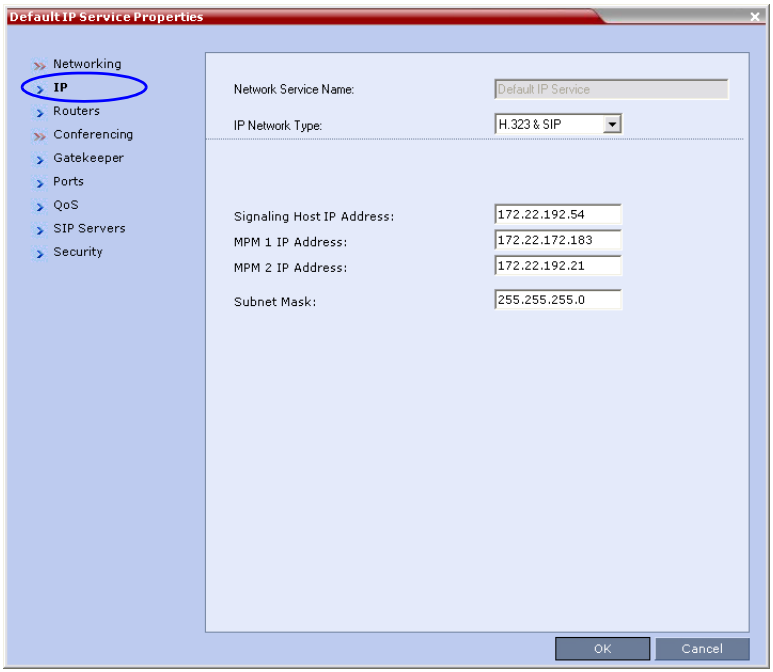
- Network type that the RMX connects to
- IP address of the RMX Signaling Host
- IP addresses of the RMX Media boards
- Subnet mask of the RMX's IP cards
- Gatekeeper parameters or add gatekeepers to the Alternate Gatekeepers list
- SIP server parameters

To view or modify the Default IP Service:

1 In the *RMX Management* pane, click **IP Network Services** ( ).

2 In the *Network* list pane, double-click the **Default IP Service** (, , or) entry.

The *Default IP Service - Networking IP* dialog box opens.



3 View or modify the following fields:

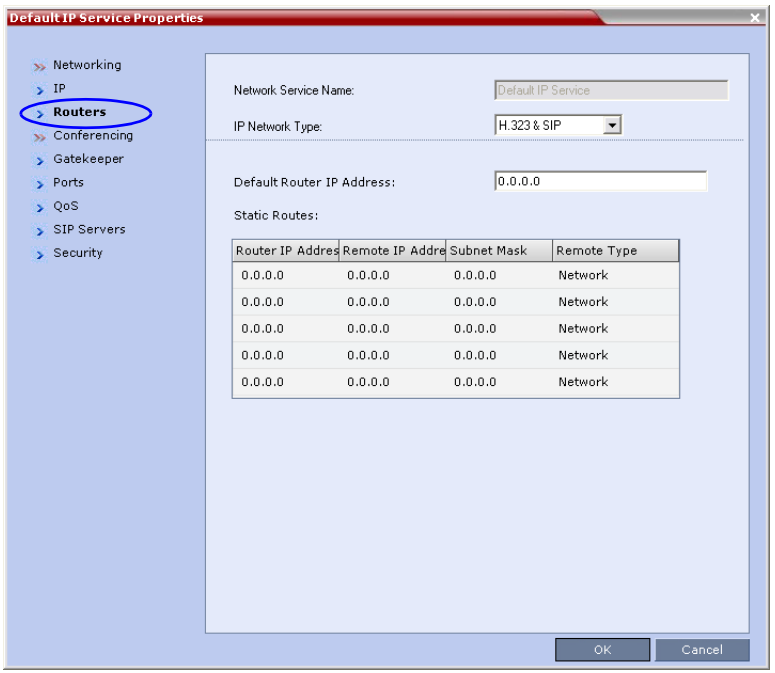
Table 12-5 *Default IP Service – Networking – IP Parameters*

Field	Description
Network Service Name	The name *Default IP Service* is assigned to the IP Network Service by the Fast Configuration Wizard. This name can be changed. **Note:** This field is displayed in all IP Signaling dialog boxes and can contain character sets that use Unicode encoding.

Table 12-5 *Default IP Service – Networking – IP Parameters (Continued)*

Field	Description
IP Network Type	Displays the network type selected during the First Entry configuration. The Default IP Network icon indicates the selected environment. You can select: • **H.323:** For an H.323-only Network Service. • **SIP:** For a SIP-only Network Service. • **H.323 & SIP:** For an integrated IP Service. Both H.323 and SIP participants can connect to the MCU using this service. **Note:** This field is displayed in all Default IP Service tabs.
Signaling Host IP Address	Enter the address to be used by IP endpoints when dialing in to the MCU. Dial out calls from the RMX are initiated from this address. This address is used to register the RMX with a Gatekeeper or a SIP Proxy server.
MPM 1 IP Address *MPM 2 IP Address*	Enter the IP address(es) of the media card (s) (MPM/MPM+ 1 and MPM/MPM+ 2 (if installed)) as provided by the network administrator.Endpoints connect to conferences and transmit call media (video, voice and content) via these addresses.
Subnet Mask	Enter the subnet mask of the MCU. Default value: 255.255.255.0.

4 If required, click the **Routers** tab.

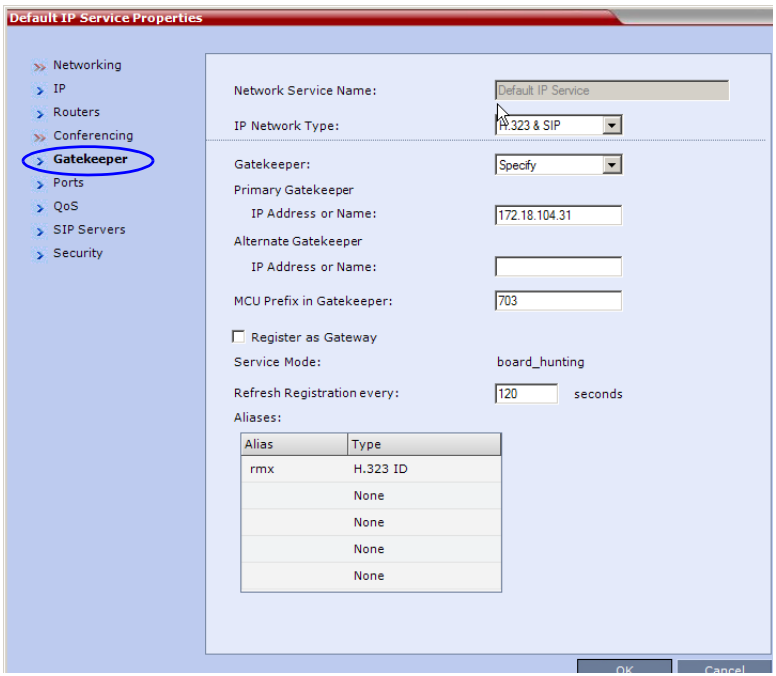


5 View or modify the fields in the *Routers* tab.

The field definitions of the *Routers* tab are the same as for the *Management Network*.

For more information, see Table 12-2 - "*Management Network Properties – Routes Parameters*" on page **12-5**.

6 If required, click the **Gatekeeper** tab.



7 View or modify the following fields:

Table 12-6 *Default IP Service – Conferencing – Gatekeeper Parameters*

Field	Description
Gatekeeper	Select **Specify** to enable configuration of the gatekeeper IP address. When **Off** is selected, all gatekeeper options are disabled.
Primary Gatekeeper IP Address or Name	Enter either the gatekeeper's host name as registered in the DNS or IP address.

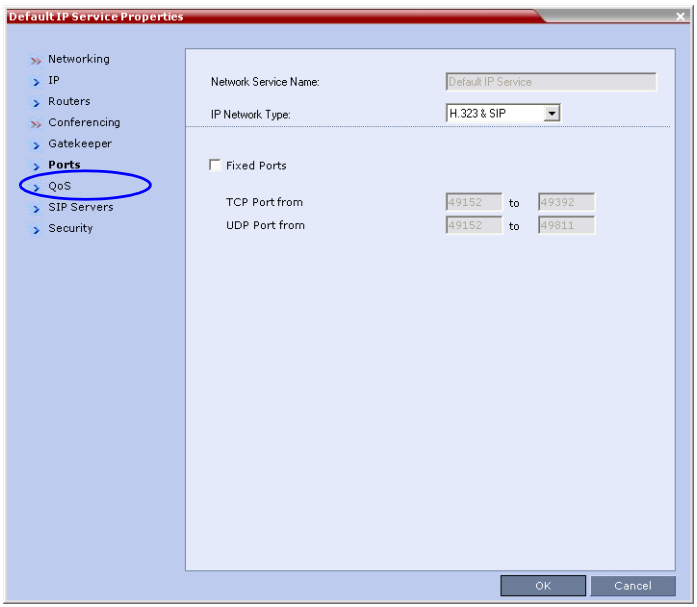
Table 12-6 *Default IP Service – Conferencing – Gatekeeper Parameters (Continued)*

Field	Description
Alternate Gatekeeper IP Address or Name	Enter the DNS host name or IP address of the gatekeeper used as a fallback gatekeeper used when the primary gatekeeper is not functioning properly.
MCU Prefix in Gatekeeper	Enter the number with which this Network Service registers in the gatekeeper. This number is used by H.323 endpoints as the first part of their dial-in string when dialing the MCU. When PathNavigator or SE200 is used, this prefix automatically registers with the gatekeeper. When another gatekeeper is used, this prefix must also be defined in the gatekeeper.
Register as Gateway	Select this check box if the RMX unit is to be seen as a gateway, for example, when using a Cisco gatekeeper. **Note:** Do not select this check box when using Polycom ReadiManager/CMA 5000 or a Radvision gatekeeper.
Refresh Registration every __ seconds	The frequency with which the system informs the gatekeeper that it is active by re-sending the IP address and aliases of the IP cards to the gatekeeper. If the IP card does not register within the defined time interval, the gatekeeper will not refer calls to this IP card until it re-registers. If set to 0, re-registration is disabled. **Note:** • It is recommended to use default settings. • This is a re-registration and not a 'keep alive' operation – an alternate gatekeeper address may be returned.

Table 12-6 *Default IP Service – Conferencing – Gatekeeper Parameters (Continued)*

Field	Description
Aliases:	
Alias	The alias that identifies the RMX's Signaling Host within the network. Up to five aliases can be defined for each RMX. **Note:** When a gatekeeper is specified, at least one prefix or alias must be entered in the table.
Type	The type defines the format in which the card's alias is sent to the gatekeeper. Each alias can be of a different type: • H.323 ID (alphanumeric ID) • E.164 (digits 0-9, * and #) • Email ID (email address format, e.g. abc@example.com) • Participant Number (digits 0-9, * and #) **Note:** Although all types are supported, the type of alias to be used depends on the gatekeeper's capabilities.

8 To view or modify fixed port settings, click the **Ports** tab.



Settings in the Ports tab allow specific ports in the firewall to be allocated to multimedia conference calls.

The port range recommended by IANA (Internet Assigned Numbers Authority) is 49152 to 65535. The MCU uses this recommendation along with the number of licensed ports to calculate the port range.

9 To modify the default settings select the **Fixed Ports** check box.

Table 12-7 *Default IP Service – Conferencing – Ports Parameters*

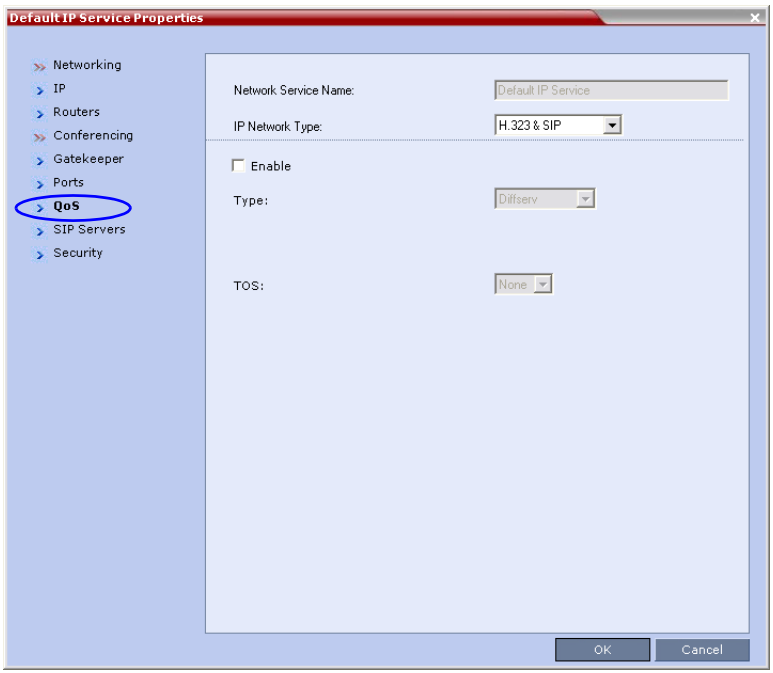
Field	Description
Fixed Ports	Leave this check box clear if you are defining a Network Service for local calls that do not require configuring the firewall to accept calls from external entities. When un-checked, the system uses the default port range. Select this option to enable other port ranges or to limit the number of ports to be left open.

Table 12-7 *Default IP Service – Conferencing – Ports Parameters*

Field	Description
TCP Port from - to	Displays the default settings for port numbers used for signaling and control. To modify the number of TCP ports, enter the first and last port numbers in the range. The number of ports is calculated as follows: Number of simultaneous calls x 2 ports (1 signaling + 1 control).
UDP Port from - to	Displays the default settings for port numbers used for audio and video. To modify the number of UDP ports, enter the first and last port numbers in the range. The number of ports is calculated as follows: Number of simultaneous calls x 6 ports (2 audio + 4 video).

If the network administrator does not specify an adequate port range, the system will accept the settings and issue a warning. Calls will be rejected when the MCU's ports are exceeded.

10 If required, click the **QoS** tab.



Quality of Service (QoS) is important when transmitting high bandwidth audio and video information. *QoS* can be measured and guaranteed in terms of:

- Average delay between packets
- Variation in delay (jitter)
- Transmission error rate

DiffServ and *Precedence* are the two *QoS* methods supported by the RMX. These methods differ in the way the packet's priority is encoded in the packet header.

RMX's implementation of *QoS* is defined per Network Service, not per endpoint.

11 View or modify the following fields:

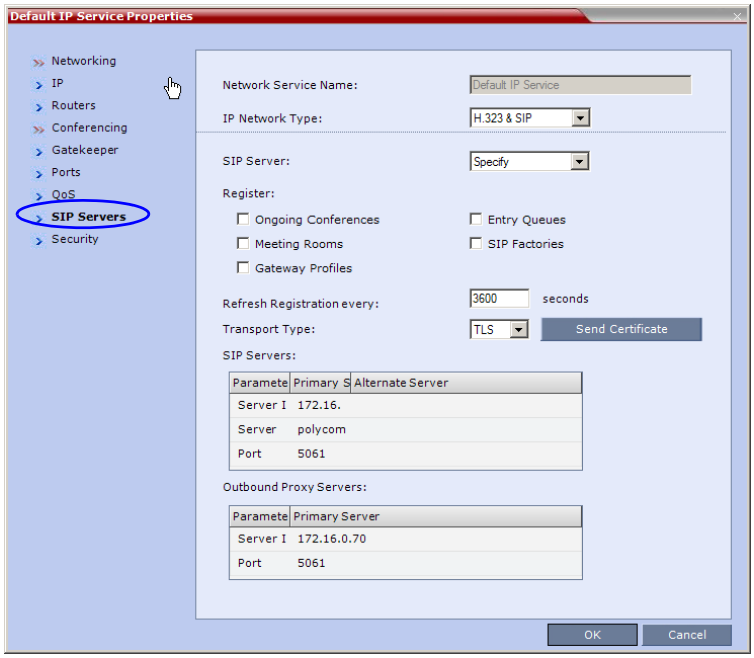
Table 12-8 *Default IP Service – Conferencing – QoS Parameters*

Field	Description
Enable	Select to enable configuration of the QoS settings. When un-checked, the system uses the default QoS settings.
Type	DiffServ and Precedence are two methods for encoding packet priority. The priority set here for audio and video packets should match the priority set in the router. • **DiffServ:** Select when the network router uses DiffServ for priority encoding. The default priority is 4 for both audio and video packets. **Note:** If you select DiffServ but your router does not support this standard, IP packets queue on the same communication links with data packets. This non-prioritized queuing greatly increases the latency and jitter in their delivery. • **Precedence:** Select when the network router uses Precedence for priority encoding, or when you are not sure which method is used by the router. Precedence should be combined with None in the TOS field. **Note:** Precedence is the default mode as it is capable of providing priority services to all types of routers, as well as being currently the most common mechanism.
Audio / Video	You can prioritize audio and video IP packets to ensure that all participants in the conference hear and see each other clearly. Select the desired priority. The scale is from 0 to 5, where 0 is the lowest priority and 5 is the highest. The recommended priority is 4 for audio and 4 for video to ensure that the delay for both packet types is the same and that audio and video packets are synchronized and to ensure lip sync.

Table 12-8 *Default IP Service – Conferencing – QoS Parameters (Continued)*

Field	Description
TOS	Select the type of Service (TOS) that defines optimization tagging for routing the conferences audio and video packets. • **Delay:** The recommended default for video conferencing; prioritized audio and video packets tagged with this definition are delivered with minimal delay (the throughput of IP packets minimizes the queue sequence and the delay between packets). • **None:** No optimization definition is applied. This is a compatibility mode in which routing is based on Precedence priority settings only. Select None if you do not know which standard your router supports.

12 To view or modify *SIP* server settings, click the **SIP Servers** tab.



13 View or modify the following fields:

Table 12-9 *Default IP Service – Conferencing – SIP Servers Parameters*

Field	Description
SIP Server	Select: • **Specify** – to manually configure SIP servers. • **Off** – if SIP servers are not present in the network.
Register: Ongoing Conferences/ Meeting Rooms/ Gateway Profiles/ Entry Queues & SIP Factories	Select the conferencing elements to register with the SIP server. Registering all the conferences and Meeting Rooms with the SIP proxy loads the proxy as the registration is constantly refreshed. It is therefore recommended to register only the Entry Queues and SIP Factories, and use the Entry Queue for conference access.
Refresh Registration every __ seconds	Enter the frequency in which the system informs the SIP proxy that it is active by re-sending the details of all registered conferencing elements to the server. If the registration is not renewed within the defined time interval, the SIP server will not refer calls to the conferencing entity until it reregisters. If timeout is set to 0, re-registration is disabled. The default value is 3600 seconds (60 minutes).

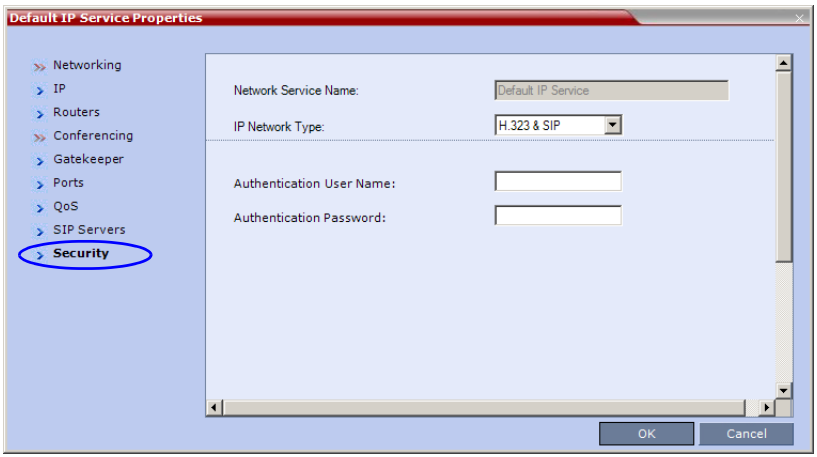
Table 12-9 *Default IP Service – Conferencing – SIP Servers Parameters (Continued)*

Field	Description
Transport Type	Select the protocol that is used for signaling between the MCU and the SIP Server or the endpoints according to the protocol supported by the SIP Server: **UDP –** Select this option to use UDP for signaling. **TCP –** Select this option to use TCP for signaling. **TLS –** The *Signaling Host* listens on secured port 5061 only and all outgoing connections are established on secured connections. Calls from SIP clients or servers to non secured ports are rejected. The following protocols are supported: TLS 1.0, SSL 2.0 and SSL 3.0. This is the required Mode for Microsoft environment. For more details, see *Appendix I, "Defining a SIP Network Service in the RMX"* on page **I-20**
SIP Servers: Primary / Alternate Server	
Server IP Address	Enter the IP address of the preferred SIP server.
Server Domain Name	Enter the name of the domain that you are using for conferences, for example: `user_name@domain name` The domain name is used for identifying the SIP server in the appropriate domain according to the host part in the dialed string. For example, when a call to `EQ1@polycom.com` reaches its outbound proxy, this proxy looks for the SIP server in the `polycom.com` domain, to which it will forward the call. When this call arrives at the SIP server in `polycom.com`, the server looks for the registered user (EQ1) and forwards the call to this Entry Queue or conference.

Table 12-9 *Default IP Service – Conferencing – SIP Servers Parameters (Continued)*

Field	Description
Port	Enter the number of the TCP or UDP port used for listening. The port number must match the port number configured in the SIP server. Default port is 5060.
Outbound Proxy Servers: Primary / Alternate Server	
Server IP Address	By default, the Outbound Proxy Server is the same as the SIP Server. If they differ, modify the IP address of the Outbound Proxy and the listening port number (if required).
Port	Enter the port number the outbound proxy is listening to. The default port is 5060.

14 To view or modify *Security* settings (SIP Digest), click the **Security** tab.



SIP Digest authentication is established by both SIP entities transmitting an MD5 (Message-Digest Algorithm 5) encrypted, shared password to each other without having to send a password in plain text over the network.

A SIP Server may require the RMX to authenticate itself during registration of SIP URIs such as Conferences, Meeting Rooms, Entry Queues and SIP factories, or when the RMX is trying to establish dial-out call via the SIP server.

15 View or modify the following fields:

Table 12-10 *Default IP Service – Security (SIP Digest)*

Field	Description
Authentication User Name	Enter the conference, Entry Queue or Meeting Room name as registered with the proxy. This field can contain up to 20 ASCII characters.
Authentication Password	Enter the conference, Entry Queue or Meeting Room password as defined in the proxy. This field can contain up to 20 ASCII characters.

If the *Authentication User Name* and *Authentication Password* fields are left empty, the SIP Digest authentication request is rejected. For registration without authentication, the RMX must be registered as a trusted entity on the SIP server.

IP Network Monitoring

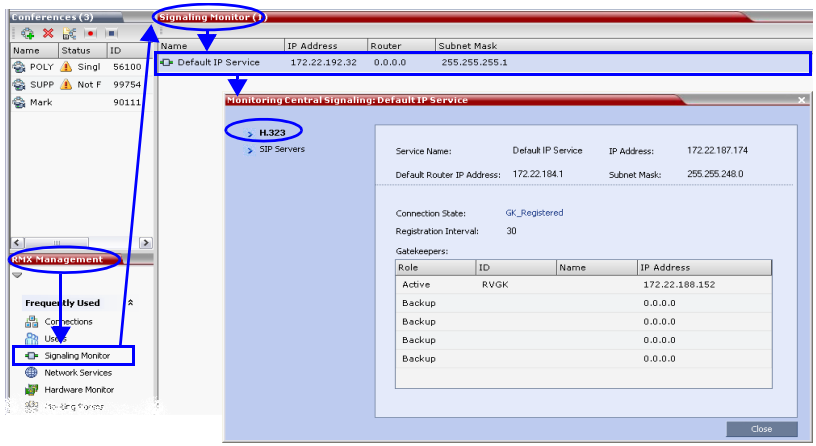
Central Signaling is the RMX entity used for monitoring the status of external network entities such as the gatekeeper, DNS, SIP proxy and Outbound proxy and their interaction with the MCU.

To monitor Central Signaling status:

1 In the *RMX Management* pane, click **Signaling Monitor** ( ).

2 In the *Central Signaling* pane, double-click **Default IP Service**.

The *Monitoring Central Signaling* dialog box opens.



The *Monitoring Central Signaling – H.323* dialog box displays the following fields:

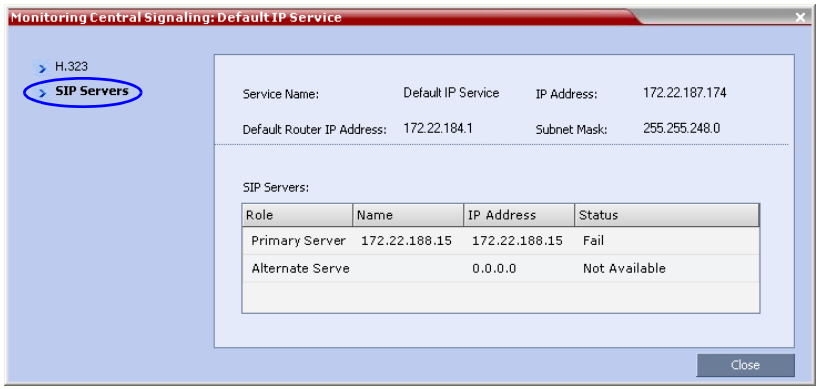
Table 12-11 *Monitoring Central Signaling – H.323 Parameters*

Field	Description
Connection State	The state of the connection between the Signaling Host and the gatekeeper: **Discovery** - The Signaling Host is attempting to locate the gatekeeper. **Registration** - The Signaling Host is in the process of registering with the gatekeeper. **Registered** - The Signaling Host is registered with the gatekeeper. **Not Registered** - The registration of the Signaling Host with the gatekeeper failed.

Table 12-11 Monitoring Central Signaling – H.323 Parameters (Continued)

Field	Description	
Registration Interval	The interval in seconds between the Signaling Host's registration messages to the gatekeeper. This value is taken from either the IP Network Service or from the gatekeeper during registration. The lesser value of the two is chosen.	
	Role	**Active** - The active gatekeeper. **Backup** - The backup gatekeeper that can be used if the connection to the preferred gatekeeper fails.
	ID	The gatekeeper ID retrieved from the gatekeeper during the registration process.
	Name	The gatekeeper's host's name.
	IP Address	The gatekeeper's IP address.

3 Click the **SIP Servers** tab.



The *Monitoring Central Signaling – SIP Servers* dialog box displays the following fields:

Table 12-12 *Monitoring Central Signaling – SIP Servers Parameters*

Field	Description
Role	**Active** -The default SIP Server is used for SIP traffic. **Backup** -The SIP Server is used for SIP traffic if the preferred proxy fails.
Name	The name of the SIP Server.
IP	The SIP Server's IP address.
Status	The connection state between the SIP Server and the Signaling Host. **Not Available** - No SIP server is available. **Auto** - Gets information from DHCP, if used.

ISDN/PSTN Network Services

To enable ISDN and PSTN participants to connect to the MCU, an ISDN/PSTN Network Service must be defined. A maximum of two ISDN/PSTN Network Services, of the same *Span Type* (E1 or T1) can be defined for the RMX. Each Network Service can attach spans from either or both cards.

Most of the parameters of the first *ISDN/PSTN Network Service* are configured in the *Fast Configuration Wizard,* which runs automatically if an RTM ISDN card is detected in the RMX during first time power-up. For more information, see the *RMX 2000 Getting Started Guide, "Procedure 3: First-time Power-up and Connection to MCU"* on page **2-9**.

Supported Capabilities and Conferencing Features:

- ISDN video is supported only in *Continuous Presence* (CP) conferences
- Only BONDING (using multiple channels as a single, large bandwidth channel) is supported
- Simple audio negotiation
- Supported video resolutions are the same as for IP
- Supported video Protocols are the same as for IP: H.261, H.263, H.264
- H.239 for content sharing
- Lecture Mode
- DTMF codes
- Securing of conferences

Non-Supported Capabilities and Conferencing Features:

- NFAS (Non-Facility Associated Signaling)
- Leased line usage
- Restricted Channel mode
- Aggregation of channels
- V.35 serial standards
- Primary and secondary clock source configuration (they are automatically selected by the system)
- Auto detection of *Audio Only* setting at endpoint
- Auto re-negotiation of bit rate
- Additional network services (two currently supported)
- Change of video mode (capabilities) from remote side during call
- Audio algorithms G.729 and G.723.1

- FECC
- H.243 Chair Control
- Encryption
- T.120 data sharing protocol
- H.261 Annex D
- Cascading using an ISDN connection as cascade link

Adding/Modifying ISDN/PSTN Network Services

You can use the *RMX Management – ISDN/PSTN Network Services* section of the *RMX Web Client* to add a second ISDN/PSTN Network Service or modify the first ISDN/PSTN Network Service.

A new ISDN/PSTN Network Service can be defined even if no RTM ISDN card is installed in the system.

Obtaining ISDN/PSTN required information

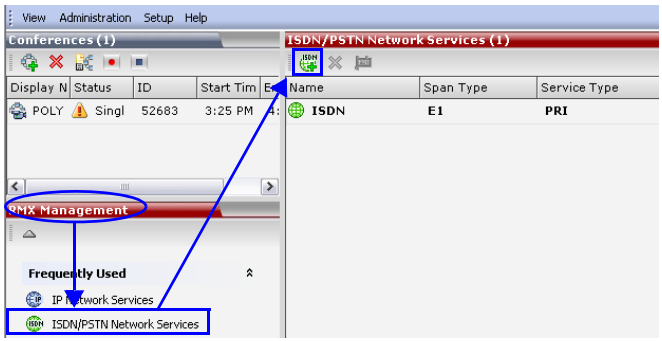
Before configuring the ISDN/PSTN Network Service, obtain the following information from your ISDN/PSTN Service Provider:

- Switch Type
- Line Coding and Framing
- Numbering Plan
- Numbering Type
- Dial-in number range

If the RMX is connected to the public ISDN Network, an external CSU or similar equipment is needed.

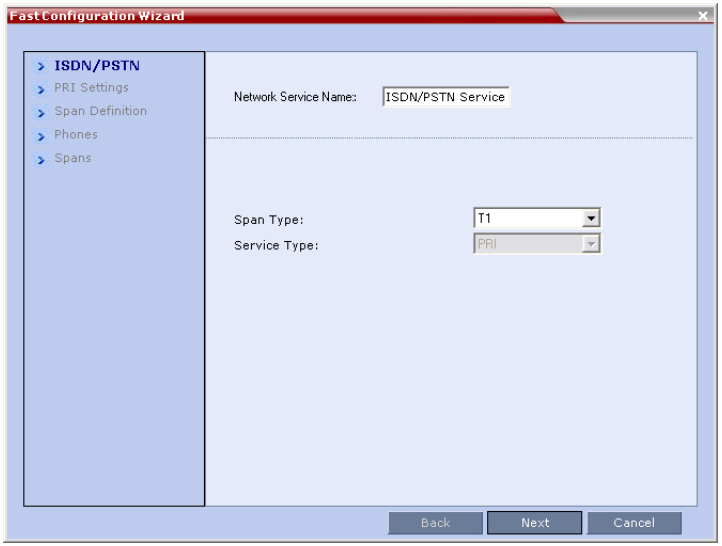
To Add an ISDN/PSTN Network Service:

1 In the *RMX Management pane*, click **ISDN/PSTN Network Services** ( ).



2 In the *ISDN/PSTN Network Services* list menu, click the **New ISDN/PSTN Service** button ( ) or right-click anywhere in the *ISDN/PSTN Network Services* list and select **New ISDN/PSTN Service**.

The *Fast Configuration Wizard* sequence begins with the *ISDN/PSTN* dialog box.



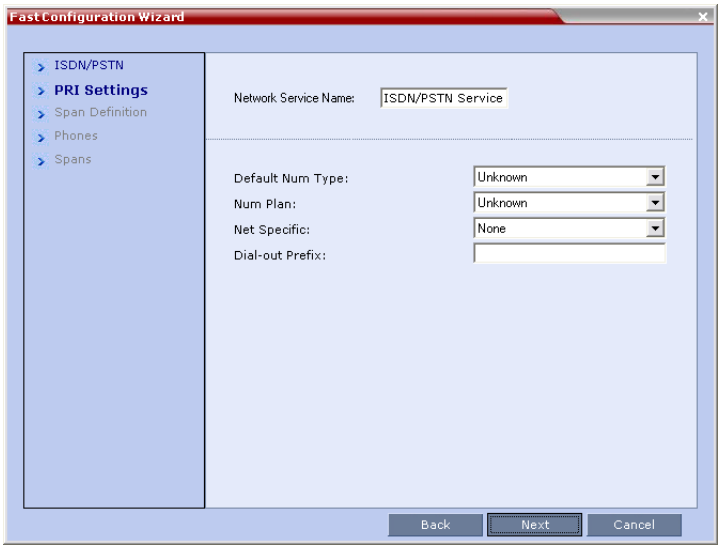
3 Define the following parameters:

Table 12-13 *ISDN Service Settings*

Field	Description
Network Service Name	Specify the service provider's (carrier) name or any other name you choose, using up to 20 characters. The Network Service Name identifies the ISDN/PSTN Service to the system. Default name: ISDN/PSTN Service **Note**: This field is displayed in all ISDN/PSTN Network Properties tabs and can contain character sets that use Unicode encoding.
Span Type	Select the type of spans (ISDN/PSTN) lines, supplied by the service provider, that are connected to the RMX. Each span can be defined as a separate Network Service, or all the spans from the same carrier can be defined as part of the same Network Service. Select either: • **T1** (U.S. – 23 B channels + 1 D channel) • **E1** (Europe – 30 B channels + 1 D channel) Default: T1
Service Type	PRI is the only supported service type. It is automatically selected.

4 Click **Next**.

The *PRI Settings* dialog box is displayed.



5 Define the following parameters:

Table 12-14 *PRI Settings*

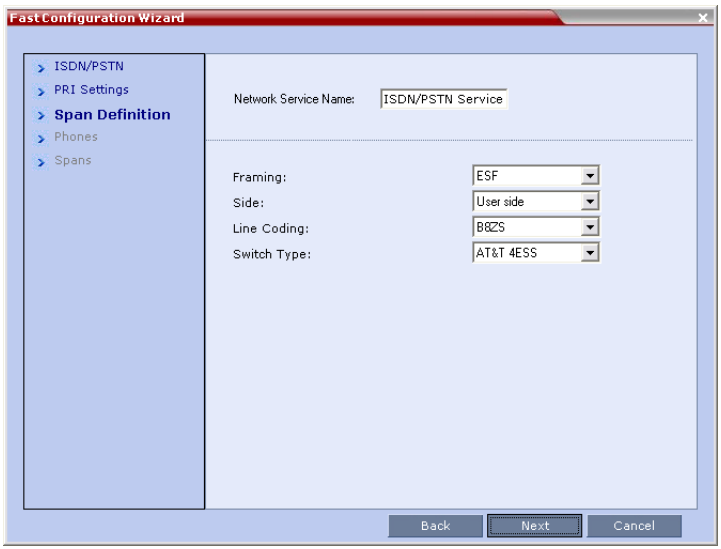
Field	Description
Default Num Type	Select the Default Num Type from the list. The Num Type defines how the system handles the dialing digits. For example, if you type eight dialing digits, the Num Type defines whether this number is national or international. If the PRI lines are connected to the RMX via a network switch, the selection of the Num Type is used to route the call to a specific PRI line. If you want the network to interpret the dialing digits for routing the call, select **Unknown**. Default: Unknown **Note:** For E1 spans, this parameter is set by the system.

Table 12-14 PRI Settings (Continued)

Field	Description
Num Plan	Select the type of signaling (Number Plan) from the list according to information given by the service provider. Default: ISDN **Note:** For E1 spans, this parameter is set by the system.
Net Specific	Select the appropriate service program if one is used by your service provider (carrier). Some service providers may have several service programs that can be used. Default: None
Dial-out Prefix	Enter the prefix that the PBX requires to dial out. Leave this field blank if a dial-out prefix is not required. The field can contain be empty (blank) or a numeric value between **0** and **9999**. Default: Blank

6 Click **Next**.

The *Span Definition* dialog box is displayed.



7 Define the following parameters:

Table 12-15 *Span Definition*

Field	Description
Framing	Select the Framing format used by the carrier for the network interface from the list. • For T1 spans, default is SFSF. • For E1 spans, default is FEBE.
Side	Select one of the following options: • User side (default) • Network side • Symmetric side **Note:** If the PBX is configured on the network side, then the RMX unit must be configured as the user side, and vice versa, or both must be configured symmetrically.

Table 12-15 *Span Definition (Continued)*

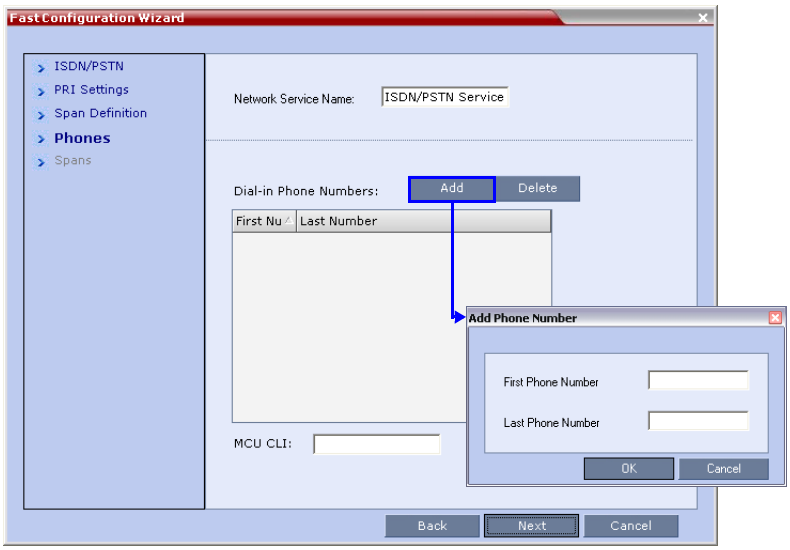
Field	Description
Line Coding	Select the PRI line coding method from the list. • For T1 spans, default is B8ZS. • For E1 spans, default is HDB3.
Switch Type	Select the brand and revision level of switch equipment installed in the service provider's central office. • For T1 spans, default is AT&T 4ESS. • For E1 spans, default is EURO ISDN.

8 Click **Next**.

The *Phones* dialog is displayed.

9 Click the **Add** button to define dial-in number ranges.

10 The *Add Phone Number* dialog box opens.



11 Define the following parameters:

Table 12-16 *Phones Settings*

Field	Description
First Number	The first number in the phone number range.
Last Number	The last number in the phone number range.

- A range must include at least two dial-in numbers.
- A range cannot exceed 1000 numbers.

12 Click **OK**.

The new range is added to the *Dial-in Phone Numbers* table.

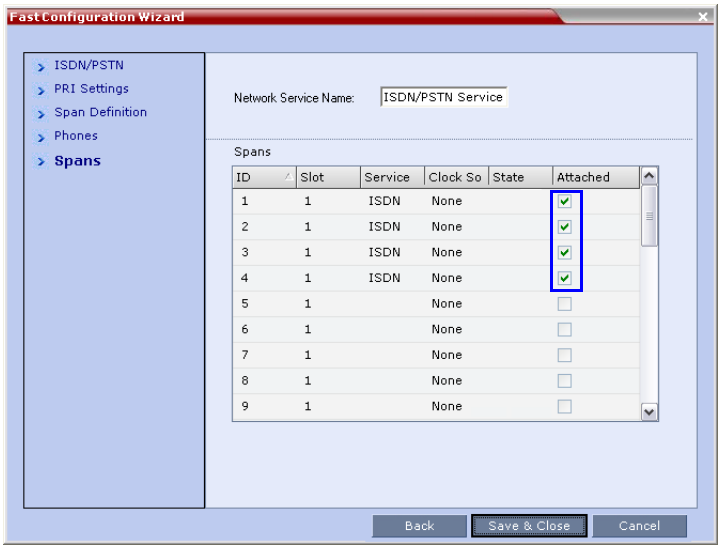
13 **Optional**. Repeat steps **9** to **10** to define additional dial-in ranges.

14 Enter the *MCU CLI* (Calling Line Identification).

In a dial-in connections, the *MCU CLI* indicates the MCU's number dialed by the participant. In a dial-out connection, indicates the MCU (CLI) number as seen by the participant

15 Click **Next**.

The *Spans* dialog box opens.



The *Spans* table displays the following read-only fields:

— ***ID*** – The connector on the ISDN/PSTN card (PRI1 - PRI12).

— ***Slot*** – The MPM/MPM+ card that the ISDN/PSTN card is connected to (1 or 2).

— ***Service*** – The Network Service to which the span is assigned, or blank if the span is not assigned to a Network Service.

— ***Clock Source*** – Indicates whether the span acts as a clock source, and if it does, whether it acts as a Primary or Backup clock source. The first span to synchronize becomes the primary clock source.

— ***State*** – The type of alarm: No alarm, yellow alarm or red alarm.

16 Assign spans to the Network Service by marking the appropriate check boxes in the ***Attached*** field.

Each ISDN/PSTN card can support up to 7 E1 or 9 T1 PRI lines.

17 Click **Save & Close**.
The new ISDN/PSTN Network Service is created and added to the *ISDN/PSTN Network Services* list.

Modifying an ISDN/PSTN Network Service

To Modify an ISDN/PSTN Network Service:

1 In the *RMX Management pane*, click the **ISDN/PSTN Network Services** ( ) icon.

2 In the *ISDN/PSTN Network Services* list, double-click the **ISDN** or right-click the **ISDN** entry and select **Properties**.

The *ISDN Properties* dialog boxes are displayed. They are similar to the *Fast Configuration Wizard's* dialog boxes. For more information see *"To Add an ISDN/PSTN Network Service:"* on page **12-29**.

The following *ISDN Properties* can be modified:

— **PRI Settings**

- *Net Specific*
- Dial-out Prefix

— **Span Definition**
  - *Framing*
  - *Side*
  - *Line Coding*
  - *Switch Type*

— **Phones**
  - *Dial-in Phone Numbers*
  - *MCU CLI*

— **Spans**
  - *Attached*

All other *ISDN Properties* can only be modified only by deleting the ISDN/PSTN network service and creating a new PSTN service using the *Fast Configuration Wizard*. For more information, see "*To Add an ISDN/PSTN Network Service:*" on page **12-29**.

13

IVR Services

Interactive Voice Response (IVR) is an application that allows participants to communicate with the conferencing system via their endpoint's input device (such as a remote control). The IVR Service includes a set of voice prompts and a video slide used to automate the participants connection to a conference or Entry Queue. It allows customization of menu driven scripts and voice prompts to meet different needs and languages.

The IVR module includes two types of services:

- Conference IVR Service that is used with conferences
- Entry Queue IVR Service that is used with Entry Queues

The system is shipped with two default Conference IVR Services (one for the conferences and the other for gateway calls) and one default Entry Queue IVR Service. The default services include voice messages and video slides in English.

To customize the IVR messages and video slide perform the following operations:

- Record the required voice messages and create a new video slide. For more information, see "*Creating a Welcome Video Slide*" on page **13-39**.
- Optional. Add the language to the list of languages supported by the system.
- Upload the voice messages to the MCU (This can be done as part of the language definition or during the IVR Service definition).
- Create the Conference IVR Service and upload the video slide, and if required any additional voice messages.
- Optional. Create the Entry Queue IVR Service and upload the required video slide and voice messages.

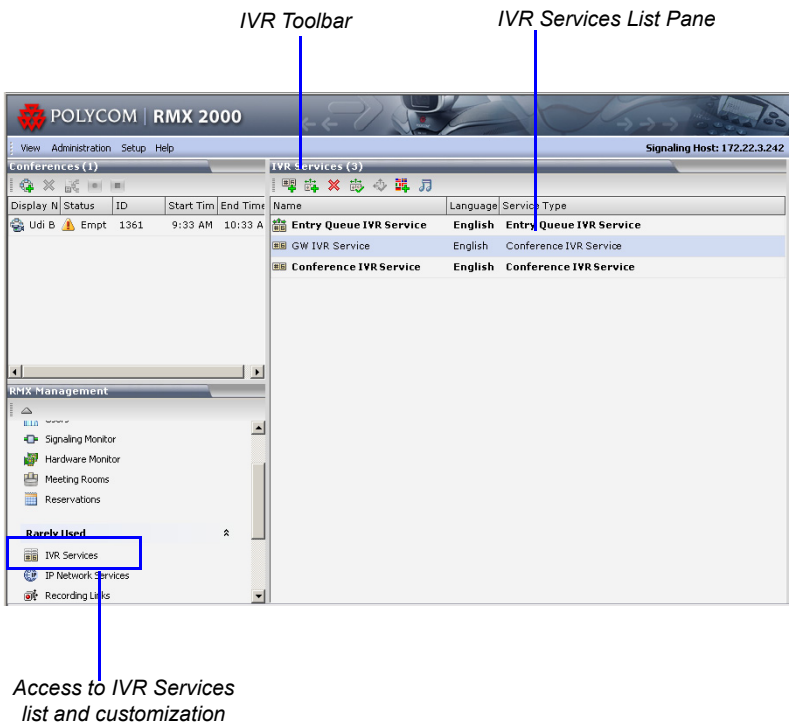
When upgrading the RMX software version new DTMF Codes and voice messages are not automatically added to existing IVR Services in order to avoid conflicts with existing DTMF codes. Therefore, to use new options, new Conference and Entry Queue IVR Services must be created.

IVR Services List

You can view the currently defined Conference IVR and Entry Queue IVR Services in the *IVR Services* list pane.

To view the IVR Services list:

1. In the *RMX Management* pane, expand the *Rarely Used* list.
2. Click the **IVR Services** ( ) entry.
   The list pane displays the *Conference IVR Services* list and the total number of IVR services currently defined in the system.



IVR Services Toolbar

The IVR Services toolbar provides quick access to the IVR Service definitions as follows:

Table 13-1 *IVR Toolbar buttons*

Button	Button Name	Descriptions
	New Conference IVR Service	To create a new Conference IVR Service.
	New Entry Queue IVR Service	To create a new Entry Queue IVR Service.
	Delete Service	Deletes the selected IVR service(s).
	Set Default Conference IVR Service	Sets the selected Conference IVR Service as default. When creating a new conference Profile the default IVR Service is automatically selected for the Profile (but can be modified).
	Set Default Entry Queue Service	Sets the selected Entry Queue IVR Service as default. When creating a new Entry Queue the default Entry Queue IVR Service is automatically selected.
	Add Supported Languages	Adds languages to the IVR module, enabling you to download voice prompts and messages for various languages.
	Replace/Change Music File	To replace the currently loaded music file that is used to play background music, the MCU is shipped with a default music file.

Adding Languages

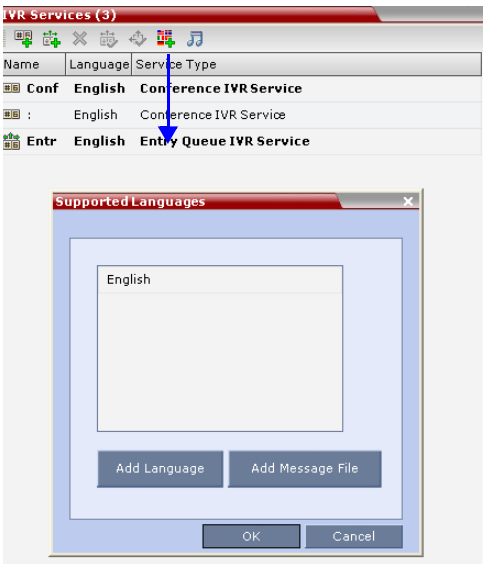
You can define different sets of audio prompts in different languages, allowing the participants to hear the messages in their preferred language.

The RMX is shipped with a default language (English) and all the prompts and messages required for the default IVR Services, conference and Entry Queues shipped with the system.

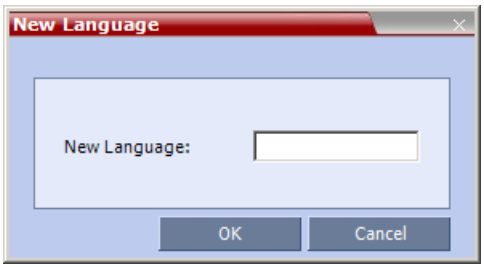
You can add languages to the list of languages for which different messages are downloaded to the MCU and IVR Services are created. This step is required before the creation of additional IVR messages using languages that are different from English, or if you want to download additional voice files to existing files in one operation and not during the IVR service definition.

To add a language:

1. In the *RMX Management* pane, expand the **Rarely Used** list.
2. Click the **IVR Services** ( ) entry.
3. In the *Conference IVR Services* list, click the **Add Supported Languages** ( ) button.
   The *Supported Languages* dialog box opens.



4 Click the **Add Language** button.
The *New Language* dialog box opens.



5 In the *New Language* box, enter the name of the new language. The language name can be typed in Unicode and cannot start with a digit. Maximum field length is 31 characters.

6 Click **OK**.
The new language is added to the list of *Supported Languages*.

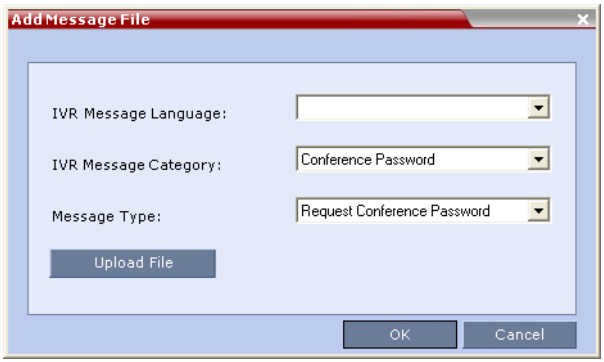
To upload messages to the MCU:

You can upload audio files for the new language or additional files for an existing language now, or you can do it during the definition of the IVR Service. In the latter case, you can skip the next steps.

- Voice messages should not exceed 3 minutes.
- It is not recommended to upload more than 1000 audio files to the MCU memory.

1 To upload the files to the MCU, in the *Supported Languages* dialog box, click the **Add Message File** button.

2 The *Add Message File* dialog box opens.



Audio files are uploaded to the MCU one-by-one.

3 In the *IVR Message Language* list, select the language for which the audio file will be uploaded to the MCU.

4 In the *IVR Message Category* list, select the category for which the audio file is uploaded.

5 In the *Message Type* list, select the message type for which the uploaded message is to be played. You can upload several audio files for each Message Type. Each file is downloaded separately.
Table 13-2 lists the Message Types for each category:

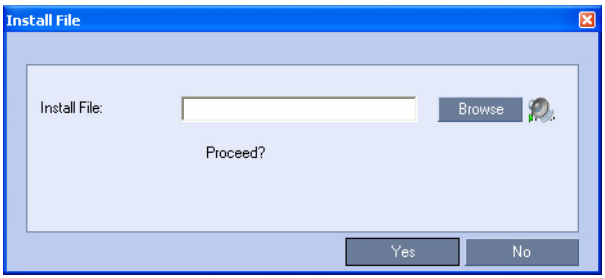
Table 13-2 *IVR Message Types by Message Category*

Message Category	Message Type	Message
Conference Password	Request Conference Password	Requests the participant to enter the conference password.
	Request Conference Password Retry	A participant who enters an incorrect password is requested to enter it again.
	Request Digit	Requests the participant to enter any digit in order to connect to the conference. Used for dial-out participants to avoid answering machines in the conference.
Welcome Message	Welcome Message	The first message played when the participant connects to the conference or Entry Queue.
Conference Chairperson	Request Chairperson Identifier	Requests the participants to enter the chairperson identifier key.
	Request Chairperson Password	Requests the participant to enter the chairperson password.
	Request Chairperson Password Retry	When the participant enters an incorrect chairperson password, requests the participant to enter it again.

Table 13-2 IVR Message Types by Message Category (Continued)

Message Category	Message Type	Message
General	Messages played for system related event notifications, for example, notification that the conference is locked. Upload the files for the voice messages that are played when an event occurs during the conference. For more information, see "*Conference IVR Service Properties - General Voice Messages*" on page 13-16.	
Billing Code	Requests the chairperson to enter the conference Billing Code.	
Roll Call	Roll call related messages, such as the message played when a participant joins the conference. Messages are listed in the *Conference IVR Service - Roll Call* dialog box.	
Conference ID	Requests the participant to enter the required Conference ID to be routed to the destination conference.	

6 Click **Upload File** to upload the appropriate audio file to the MCU. The *Install File* dialog box opens.



7 Enter the file name or click the **Browse** button to select the audio file to upload.
The *Select Source File* dialog box opens.

8 Select the appropriate *.wav audio file, and then click the **Open** button.
The name of the selected file is displayed in the *Install* field in the *Install File* dialog box.

9 Optional. You can play a .wav file by selecting the *Play* button ( ).

10 Click **Yes** to upload the file to the MCU.
The system returns to the *Add Message File* dialog box.

11 Repeat step 6 to 10 for each additional audio file to be uploaded to the MCU.

12 Once all the audio files are uploaded to the MCU, close the *Add Message File* dialog box and return to the *Add Language* dialog box.

13 Click **OK**.

Defining a New Conference IVR Service

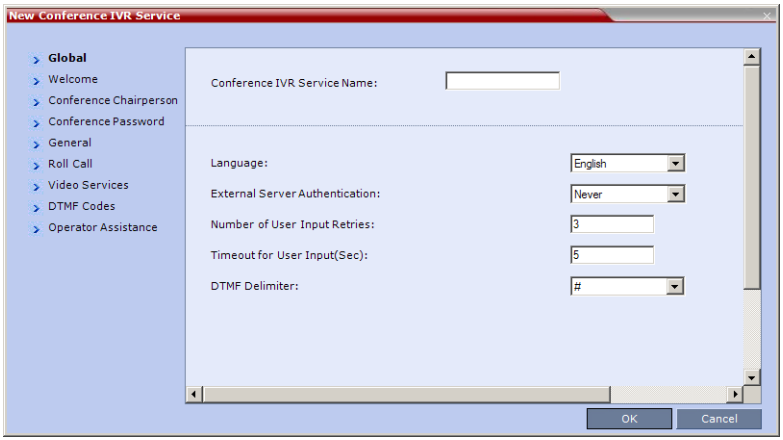
The RMX is shipped with two default Conference IVR Services and all its audio messages and video slide. You can define new Conference IVR Services or modify the default Conference IVR Service. For the definition of Conference IVR Service for gateway calls, see *"Defining the IVR Service for Gateway Calls"* on page **15-12**.

Up to 40 IVR Services (Conference IVR Services and Entry Queue IVR Services) can be defined for a single RMX unit.

Defining a New Conference IVR Service

To define a new Conference IVR Service:

1 On the *IVR Services* toolbar, click the **New Conference IVR Service** ( ) button.
The *New Conference IVR Service - Global* dialog box opens.



2 Define the following parameters:

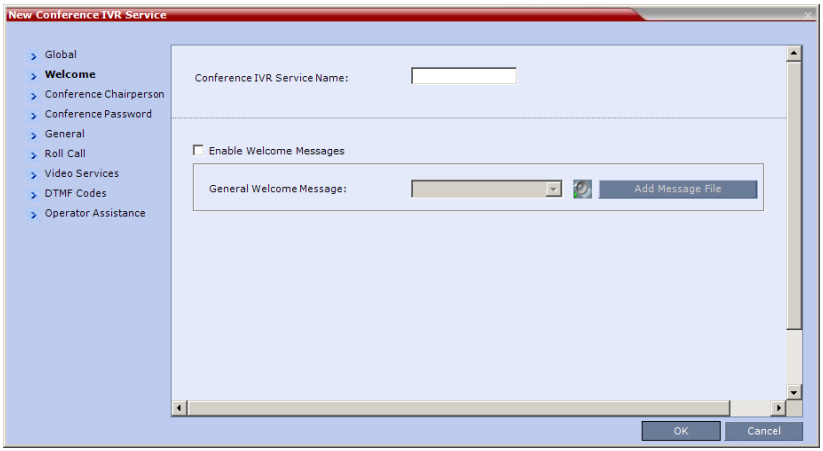
Table 13-3 *Conference IVR Service Properties - Global Parameters*

Field/Option	Description
Conference IVR Service Name	Enter the name of the Conference IVR Service. The maximum field length is 20 characters and may be typed in Unicode.
Language For IVR	Select the language of the audio messages and prompts from the list of languages defined in the *Supported languages*. The default language is English. For more information, see "*Adding Languages"* on page **13-4**.
External Server Authentication	You can configure the IVR Service to use an external database application to verify a participant's right to join the conference. For more information, see *Appendix D:* "*Conference Access with External Database Authentication"* on page **D-6**. Select one of the following options: • ***Never*** – The participant's right to join the conference will not be verified with an external database application (default). • ***Always*** – Any participant request to join the conference is validated with the external database application using a password. • ***Upon Request*** – Only the participant request to join the conference as chairperson is validated with the external database application using a password. The validation process occurs only when the participant enters the chairperson identifier key.
Number of User Input Retries	Enter the number of times the participant will be able to respond to each menu prompt before being disconnected from the conference. Range is between 1-4, and the default is 3.
Timeout for User Input (Sec)	Enter the duration in seconds that the system will wait for the participant's input before prompting for another input. Range is between 1-10, and the default value is 5 seconds.

Table 13-3 *Conference IVR Service Properties - Global Parameters (Continued)*

Field/Option	Description
DTMF Delimiter	Enter the key that indicates the last input key. Possible values are the pound (#) and star (*) keys. The default is #.

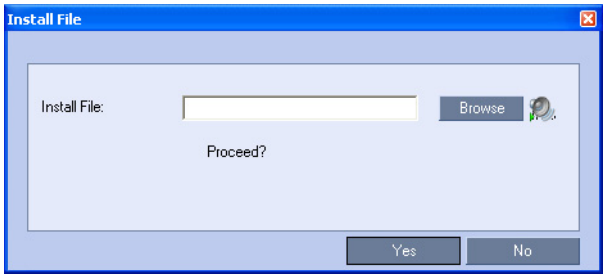
3 Click the **Welcome** tab.
The *New Conference IVR Service - Welcome* dialog box opens.



4 Select the **Enable Welcome Messages** check box to define the system behavior when the participant enters the Conference IVR queue. When participants access a conference through an Entry Queue, they hear messages included in both the Entry Queue Service and Conference IVR Service. To avoid playing the Welcome Message twice, disable the Welcome Message in the Conference IVR Service.

5 Select the **General Welcome Message,** to be played when the participant enters the conference IVR queue.

6 To upload an audio file for an IVR message, click **Add Message File**. The *Install File* dialog box opens.



The RMX unit is bundled with default audio IVR message files. To upload a customized audio file, see *"Creating Audio Prompts and Video Slides"* on page 13-35.

a Click the **Browse** button to select the audio file (*.wav) to upload. The *Select Source File* dialog box opens.

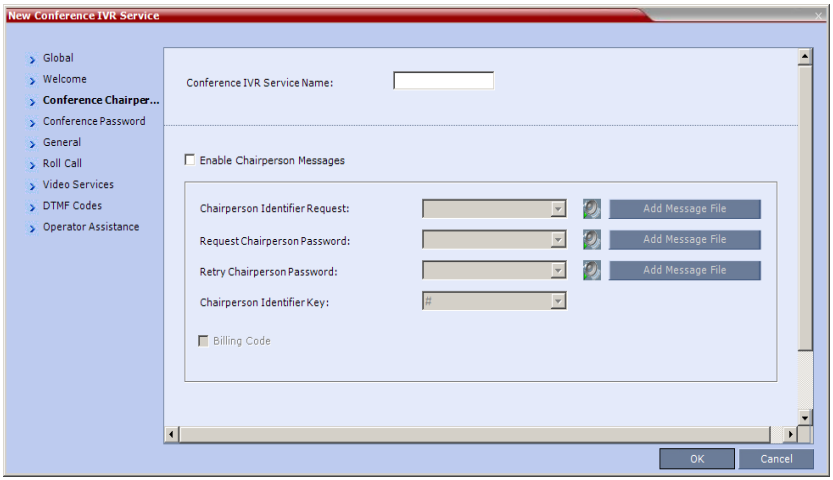
b Select the appropriate *.wav audio file and then click the **Open** button.

c Optional. You can play a .wav file by selecting the *Play* button ( ).

d In the *Install File* dialog box, click **Yes** to upload the file to the MCU memory.
The *Done* dialog box opens.

e Once the upload is complete, click **OK** and return to the *IVR* dialog box. The new audio file can now be selected from the list of audio messages.

7 Click the **Conference Chairperson** tab.
The *New Conference IVR Service - Conference Chairperson* dialog box opens.



8 Select the **Enable Chairperson Messages** check box to enable the chairperson functionality. If this feature is disabled, participants are not able to connect as the chairperson.

9 Select the various voice messages and options for the chairperson connection.

If the files were not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the RMX.

Table 13-4 *New Conference IVR Service Properties - Conference Chairperson Options and Messages*

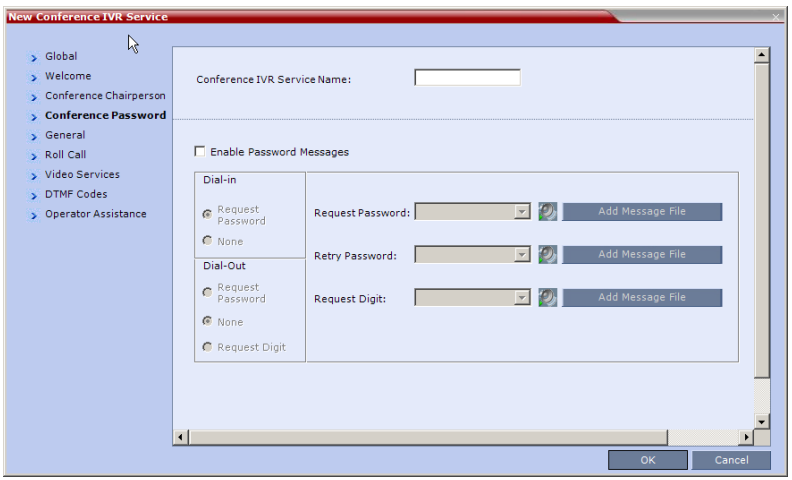
Field/Option	Description
Chairperson Identifier Request	Select the audio file that requests the participants to enter the key that identifies them as the conference chairperson.
Request Chairperson Password	Select the audio file that prompts the participant for the chairperson password.

Table 13-4 New Conference IVR Service Properties - Conference Chairperson Options and Messages (Continued)

Field/Option	Description
Retry Chairperson Password	Select the audio file that prompts participants to re-enter the chairperson password if they enter it incorrectly.
Chairperson Identifier Key	Enter the key to be used for identifying the participant as a chairperson. Possible keys are: pound key (#) or star (*).
Billing Code	The prompt requesting the chairperson billing code selected in the General tab.

10 Click the **Conference Password** tab.

The *New Conference IVR Service - Conference Password* dialog box opens.



11 Select the **Enable Password Messages** check box to request the conference password before moving the participant from the conference IVR queue to the conference.

12 Select the MCU behavior for password request for *Dial-in* and *Dial-out* participant connections.

Select the required system behavior as follows:

— **Request password** - The system requests the participant to enter the conference password.

— **None** - The participant is moved to the conference without any password request.

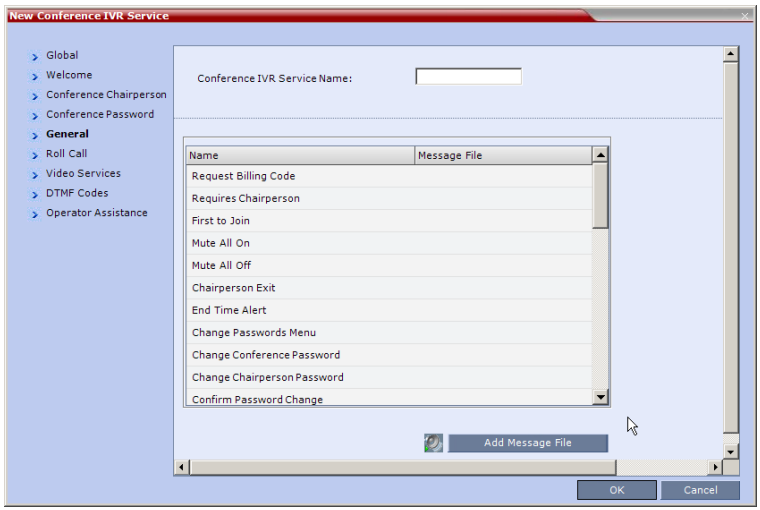
— **Request Digit** - The system requests the participant to enter any key. This option is used mainly for dial-out participants and to prevent an answering machine from entering the conference.

13 Select the various audio messages that will be played in each case.

Table 13-5 *New Conference IVR Service Properties - Conference Password Parameters*

Option	Description
Request Password	Select the audio file that prompts the participant for the conference password.
Retry Password	Select the audio file that requests the participant to enter the conference password again when failing to enter the correct password.
Request Digit	Select the audio file that prompts the participant to press any key when the *Request Digit* option is selected.

14 Click the **General** tab.
The *New Conference IVR Service - General* dialog box opens.



The *General* dialog box lists messages that are played during the conference. These messages are played when participants or the conference chairperson perform various operations or when a change occurs.

15 To assign the appropriate audio file to the message type, click the appropriate table entry, in the *Message File* column. A drop-down list is enabled.

16 From the list, select the audio file to be assigned to the event/ indication.

17 Repeat steps 15 and 16 to select the audio files for the required messages.
The following types of messages and prompts can be enabled:

Table 13-6 *Conference IVR Service Properties - General Voice Messages*

Message Type	Description
Request Billing Code	Requests the participant to enter a code for billing purposes.

Table 13-6 *Conference IVR Service Properties - General Voice Messages (Continued)*

Message Type	Description
Requires Chairperson	The message is played when the conference is on hold and the chairperson joins the conference. For this message to be played the *Conference Requires Chairperson* option must be selected in the *Conference Profile - IVR* dialog box.
First to Join	Informs the participant that he or she is the first person to join the conference.
Mute All On	Informs all participants that they are muted, with the exception of the conference chairperson. **Note:** This message is played only when the *Mute All Except Me* option is activated.
Mute All Off	This message is played to the conference to inform all participants that they are unmuted (when *Mute All* is cancelled).
Chairperson Exit	Informs all the conference participants that the chairperson has left the conference, causing the conference to automatically terminate after a short interval. **Note:** This message is played only when the *Requires Chairperson* option is selected in the *Conference Profile - IVR* dialog box.
End Time Alert	Indicates that the conference is about to end.
Change Passwords Menu	This voice menu is played when the participants requests to change the conference password. This message details the steps required to complete the procedure.
Change Conference Password	Requests the participant to enter a new conference password when the participant is attempting to modify the conference password.
Change Chairperson Password	Requests the participant to enter a new chairperson password when the participant is attempting to modify the chairperson password.

Table 13-6 *Conference IVR Service Properties - General Voice Messages (Continued)*

Message Type	Description
Confirm Password Change	Requests the participant to re-enter the new password.
Change Password Failure	A message played when the participant enters an invalid password, for example when a password is already in use.
Password Changed Successfully	A message is played when the password was successfully changed.
Self Mute	A confirmation message that is played when participants request to mute their line.
Self Unmute	A confirmation message that is played when participants request to unmute their line.
Chairperson Help Menu	A voice menu is played upon a request from the chairperson, listing the operations and their respective DTMF codes that can be performed by the chairperson. The playback can be stopped any time. **Note:** If you modify the default DTMF codes used to perform various operations, the default voice files for the help menus must be replaced.
Participant Help Menu	A voice menu that is played upon request from a participant, listing the operations and their DTMF codes that can be performed by any participant.
Maximum Number of Participants Exceeded	Indicates the participant cannot join the destination conference as the maximum allowed number of participants will be exceeded.
Recording in Progress	This message is played to participant joining a conference that is being recorded indicating the recording status of the conference.
Recording Failed	This message is played when the conference recording initiated by the chairperson or the participant (depending on the configuration) fails to start.

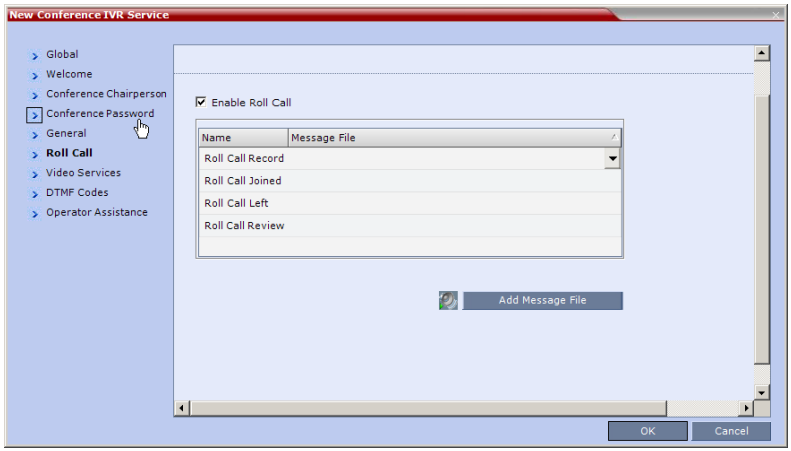
Table 13-6 *Conference IVR Service Properties - General Voice Messages (Continued)*

Message Type	Description
Conference is Secured	This message is played when the conference status changes to Secure as initiated by the conference chairperson or participant (using DTMF code *71).
Conference is unsecured	This message is played when the conference status changes to Unsecured as initiated by the conference chairperson or participant (using DTMF code #71).
Conference is Locked	This message is played to participants attempting to join a Secured conference.
Enter Destination ID	Prompts the calling participant for the destination number. Default message prompts the participant for the conference ID (same message as in the Entry Queue IVR Service).
Incorrect Destination ID	If the participant entered an incorrect conference ID (in gateway calls it is the destination number), requests the participant to enter the number again.
Dial Tone	The tone that will be played to indicate a dialing tone, to let the calling participant enter the destination number.
Ringing Tone	The tone that will be played to indicate that the system is calling the destination number.

18 Click the **Roll Call** tab.
The *New Conference IVR Service - Roll Call* dialog box opens.
The Roll Call feature of the Conference IVR Service is used to record the participants' names for playback when the participants join and leave a conference.

When the system flag IVR_ROLL_CALL_USE_TONES_INSTEAD_OF_VOICE is set to YES, the system does not playback the Roll Call names when participants enter or exit the conference. If the voice messages are replaced with tones, the system will play these tones instead.

19 To enable the Roll Call feature, select the **Enable Roll Call** check box.



20 To assign the audio file to the message type, in the Message File column, click the appropriate table entry. An arrow appears in the *Message File* column.

If the Roll Call option is enabled, you must assign the appropriate audio files to all message types.

21 Click the arrow to open the *Message File* list and select the appropriate audio file.

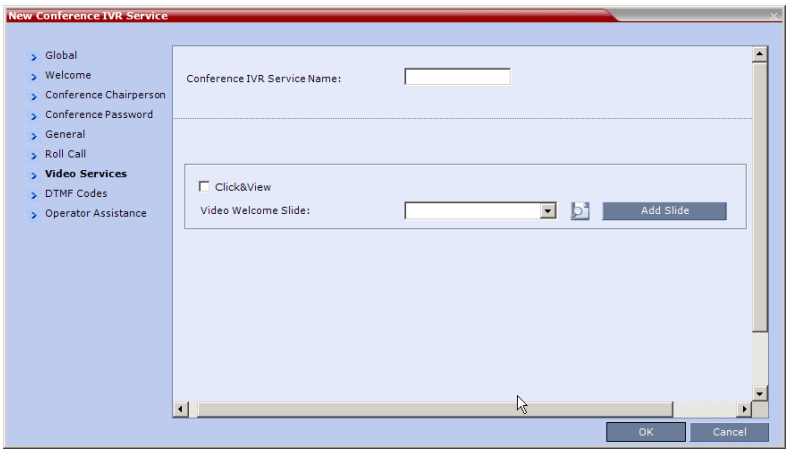
Table 13-7 *Conference IVR Service Properties - Roll Call Messages*

Roll Call Message	Description
Roll Call Record	Requests participants to state their name for recording, when they connect to the conference. **Note:** The recording is automatically terminated after two seconds.

Table 13-7 *Conference IVR Service Properties - Roll Call Messages (Continued)*

Roll Call Message	Description
Roll Call Joined	A voice message stating that the participant has joined the conference. **Note:** When the system flag *IVR_ROLL_CALL_USE_TONES_INSTEAD_OF_VOICE* is set to **YES**, the system does not playback the Roll Call names when participants enter the conference. However, the voice message will be played, unless it is replaced with tone file. The use of tones requires the uploading of the appropriate tone files in *wav format and replacing the Roll Call Joined message file with the tone file.
Roll Call Left	A voice message stating that the participant has left the conference. **Note:** When the system flag *IVR_ROLL_CALL_USE_TONES_INSTEAD_OF_VOICE* is set to **YES**, the system does not playback the Roll Call names when participants exit the conference. However, the voice message will be played, unless it is replaced with tone file. The use of tones requires the uploading of the appropriate tone files in *wav format and replacing the Roll Call Left message file with the tone file.
Roll Call Review	Played when Roll Call is requested by the chairperson, introducing the names of the conference participants in the order they joined the conference.

22 Click the **Video Services** tab.
The *New Conference IVR Service - Video Services* dialog box opens.



23 Define the following parameters:

Table 13-8 *New Conference IVR Service Properties - Video Services Parameters*

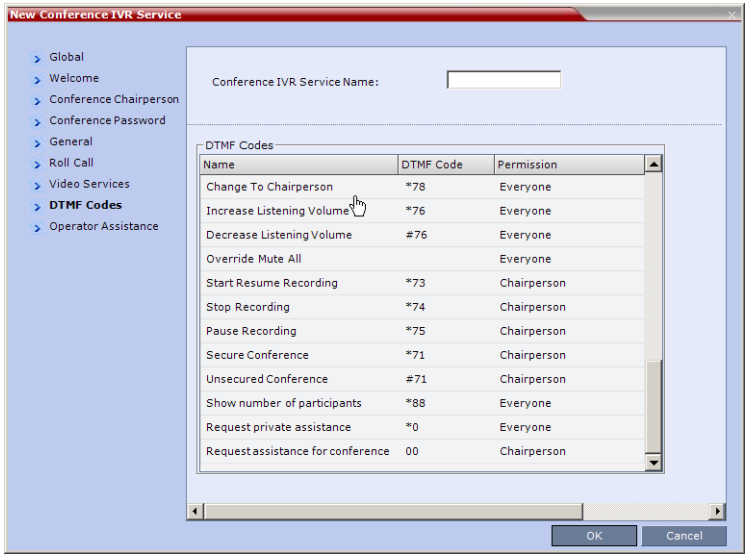
Video Services	Description
Click&View	Select this option to enable endpoints to run the Click&View application that enables participants to select a video layout from their endpoint.

Table 13-8 *New Conference IVR Service Properties - Video Services Parameters (Continued)*

Video Services	Description
Video Welcome Slide	Select the video slide file to be displayed when participants connect to the conference. To view any slide, click the **Preview Slide** ( ) button. If the video slide file was not uploaded to the MCU prior to the IVR Service definition, click the **Add Slide** button. The *Install File* dialog box opens. The uploading process is similar to the uploading of audio files. For more information, see step 6 on page 13-11. **Notes:** • When using one of the default Polycom slides, the slide will be displayed in the resolution defined in the profile, i.e. CIF, SD, HD 720p or HD 1080p. When using a custom slide, it will be displayed in the only in CIF resolution. • When defining a gateway IVR Service, the recommended default slide is: Default_GW_Welcome_Slide.

24 Click the **DTMF Codes** tab.

The *New Conference IVR Service - DTMF Codes* dialog box opens.



This dialog box lists the default DTMF codes for the various functions that can be performed during the conference by all participants or by the chairperson.

Table 13-9 *New Conference IVR Service Properties - DTMF Codes*

Operation	DTMF String	Permission
Mute My Line	*6	All
Unmute My Line	#6	All
Increase Broadcast Volume	*9	All
Decrease Broadcast Volume	#9	All
Mute All Except Me	*5	Chairperson
Cancel Mute All Except Me	#5	Chairperson
Change Password	*77	Chairperson
Mute Incoming Participants	*86	Chairperson
Unmute Incoming Participants	#86	Chairperson

Table 13-9 *New Conference IVR Service Properties - DTMF Codes*

Operation	DTMF String	Permission
Play Help Menu	*83	All
Enable Roll Call	*32	Chairperson
Disable Roll Call	#32	Chairperson
Roll Call Review Names	*33	Chairperson
Roll Call Stop Review Names	#33	Chairperson
Terminate Conference	*87	Chairperson
Start Click&View	**	All
Change To Chairperson	*78	All
Increase Listening Volume	*76	All
Decrease Listening Volume	#76	All
Override Mute All	Configurable	All
Start Recording	*73	Chairperson
Stop Recording	*74	Chairperson
Pause Recording	*75	Chairperson
Secure Conference	*71	Chairperson
Unsecured Conference	#71	Chairperson
Show Number of Participants	*88	All
Request individual assistance	*0	All
Request assistance for conference	00	Chairperson

25 To modify the DTMF code or permission:

a In the *DTMF Code* column, in the appropriate entry enter the new code.

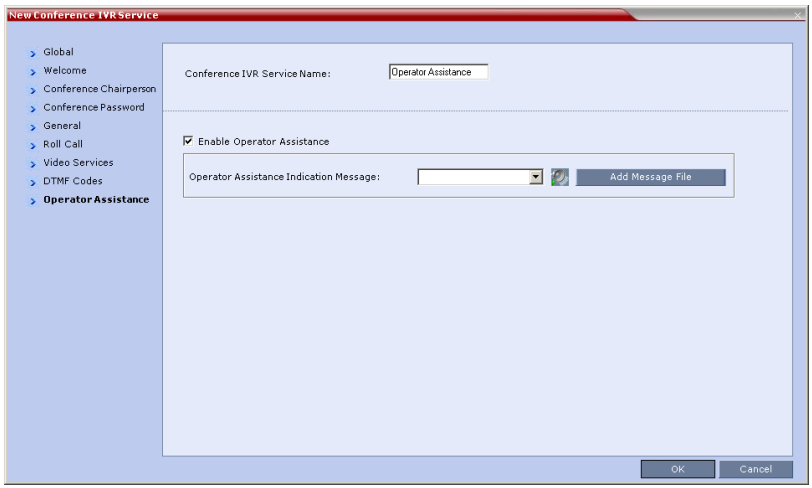
b In the *Permission* column, select from the list who can use this feature (all or just the chairperson).

By default, the Secure, Unsecure Conference and Show Number of Participants options are enabled in the Conference IVR Service. These options can be disabled and must be disabled removing their codes from the Conference IVR Service.

- To disable the Secure Conference options, in the *DTMF Code* column, clear the DTMF codes of both Secured Conference **(*71)** and Unsecured Conference **(#71)** from the table.
- To disable the Text Indication option in the DTMF Code column, clear the DTMF code **(*88)** of *Show Number of Participants* from the table.

26 Click the **Operator Assistance** tab.

The *Operator Assistance* dialog box opens.



27 Select **Enable Operator Assistance** to enable operator assistance when the participant requires or requests help during the connection process to the conference or during the conference.

28 In the *Operator Assistance Indication Message* field, select the audio message to be played when the participant requests or is waiting for the operator's assistance.

If the audio file was not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the RMX.

29 Click **OK** to complete the IVR Service definition.
The new Conference IVR Service is added to the *IVR Services* list.

Entry Queues IVR Service

An Entry Queue (EQ) is a routing lobby for conferences. Participants are routed to the appropriate conference according to the conference ID they enter.

An Entry Queue IVR Service must be assigned to the Entry Queue to enable the voice prompts and video slide guiding the participants through the connection process.

An Entry Queue IVR Service is a subset of an IVR Service. You can create different Entry Queue Services for different languages and personalized voice messages.

The RMX is shipped with a default Entry Queue IVR Service and all its audio messages and video slide. You can define new Entry Queue IVR Services or modify the default Entry Queue IVR Service.

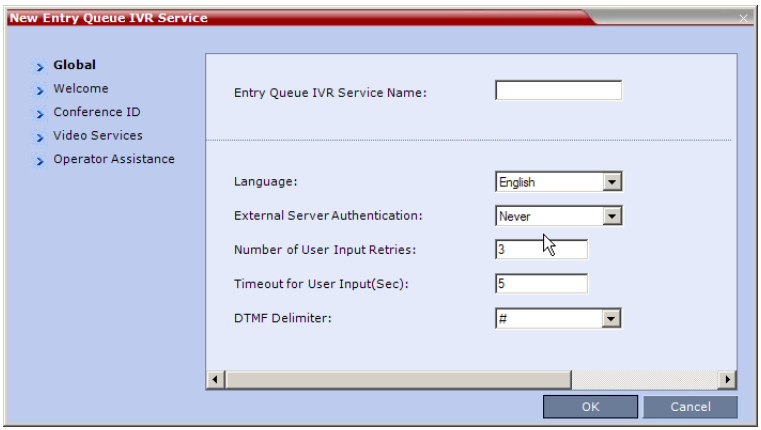
Defining a New Entry Queue IVR Service

To set up a new Entry Queue IVR Service:

1 In the *RMX Management* pane, click **IVR Services** ( ).

2 In the *IVR Services* list, click the **New Entry Queue IVR Service** ( ) button.

The *New Entry Queue IVR Service - Global* dialog box opens.

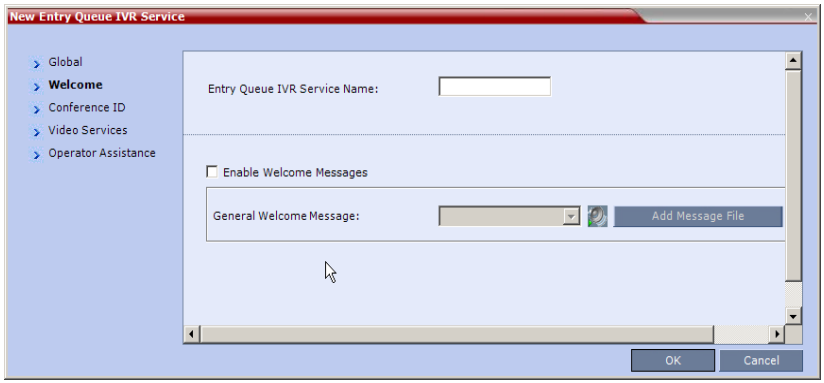


3 Fill in the following parameters:

Table 13-10 Entry Queue IVR Service Properties - Global Parameters

Option	Description
Entry Queue Service Name	(Mandatory) Enter the name of the Entry Queue Service. The name can be typed in Unicode. Maximum field length is 80 ASCII characters.
Language	Select the language in which the Audio Messages and prompts will be heard. The languages are defined in the *Supported Languages* function.
External Server Authentication	This option is used for Ad Hoc conferencing, to verify the participant's right to initiate a new conference. For a detailed description see *Appendix D:* "*Conference Access with External Database Authentication"* on page **D-6**. Select one of the following options: • **None** to start a new conference without verifying with an external database the user right to start it. • **Conference ID** to verify the user's right to start a new conference with an external database application using the conference ID.
Number of User Input Retries	Enter the number of times the participant is able to respond to each menu prompt before the participant is disconnected from the MCU.
Timeout for User Input (Sec.)	Enter the duration in seconds that the system waits for input from the participant before it is considered as an input error.
DTMF Delimiter	The interaction between the caller and the system is done via touch-tone signals (DTMF codes). Enter the key that will be used to indicate a DTMF command sent by the participant or the conference chairperson. Possible keys are the pound key (#) or star (*).

4 Click the **Welcome** tab.
The *New Entry Queue IVR Service - Welcome* dialog box opens.

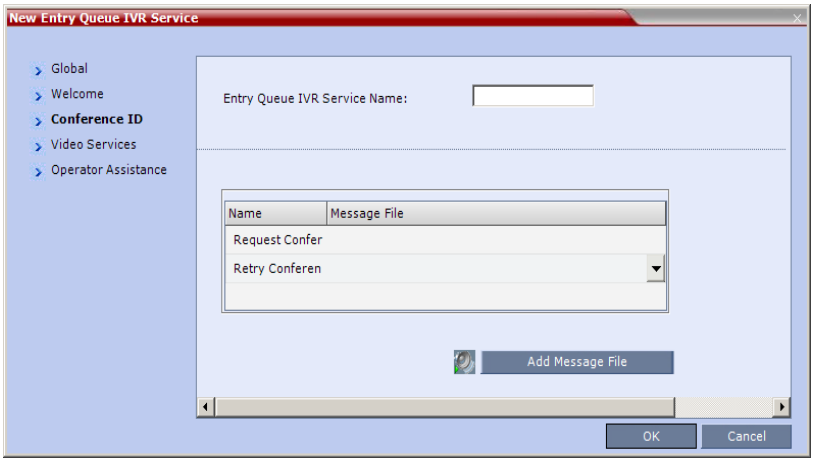


If the files were not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the RMX.

5 Define the appropriate parameters. This dialog box contains options that are identical to those in the *Conference IVR Service - Welcome Message* dialog box. For more information about these parameters, see Table 13-4 on page **13-13**.

6 Click the **Conference ID** tab.

The *New Entry Queue IVR Service - Conference ID* dialog box opens.



7 Select the voice messages:

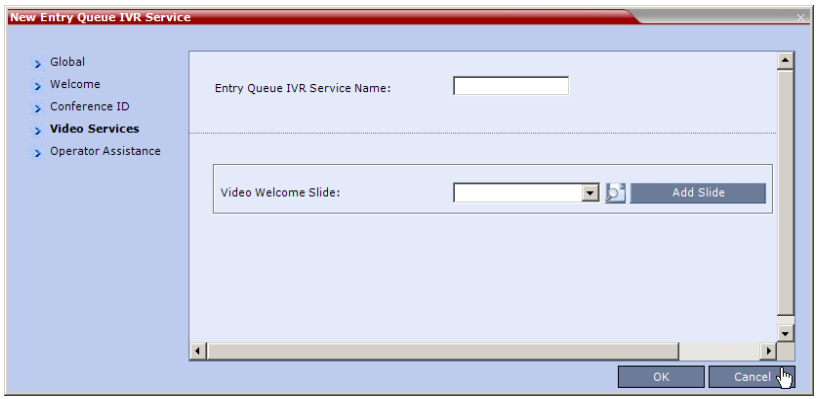
Table 13-11 Entry Queue IVR Service Properties - Conference ID

Field/Option	Description
Request Conference ID	Prompts the participant for the conference ID.
Retry Conference ID	When the participant entered an incorrect conference ID, requests the participant to enter the ID again.

8 Assign an audio file to each message type, as follows:

— In the *Message File* column, click the table entry, and then select the appropriate audio message.

9 Click the **Video Services** tab.
The *New Entry Queue IVR Service - Video Services* dialog box opens.



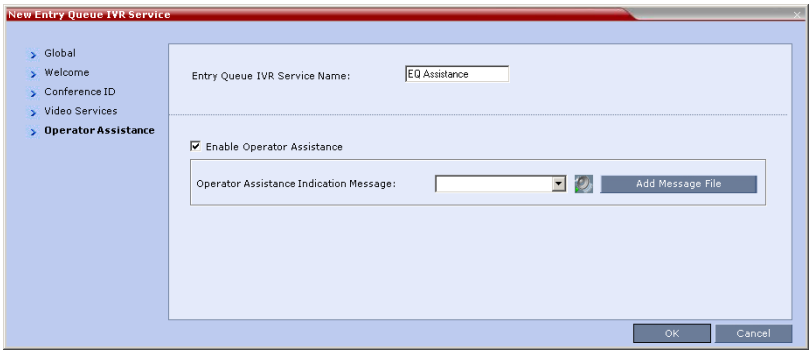
10 In the *Video Welcome Slide* list, select the video slide that will be displayed to participants connecting to the Entry Queue. The slide list includes the video slides that were previously uploaded to the MCU memory.

11 To view any slide, click the **Preview Slide** ( ) button. If the list is empty, you can upload a new slide by clicking the **Add Slide** button.

The *Install File* dialog box opens. The uploading process is similar to the uploading of audio files, see step 6 on page **13-11**.

The video slide must be in a .jpg or .bmp file format. For more information, see "*Creating a Welcome Video Slide"* on page **13-39**.

12 Click the **Operator Assistance** tab.
The *Operator Assistance* dialog box opens.



13 Select **Enable Operator Assistance** to enable operator assistance when the participant requires or requests help during the connection process.

14 In the *Operator Assistance Indication Message* field, select the audio message to be played when the participant requests or is waiting for operator's assistance.

If the audio file was not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the RMX.

15 Click **OK** to complete the Entry Queue Service definition.
The new Entry Queue IVR Service is added to the *IVR Services* list. For more information, see "*IVR Services List"* on page **13-2**.

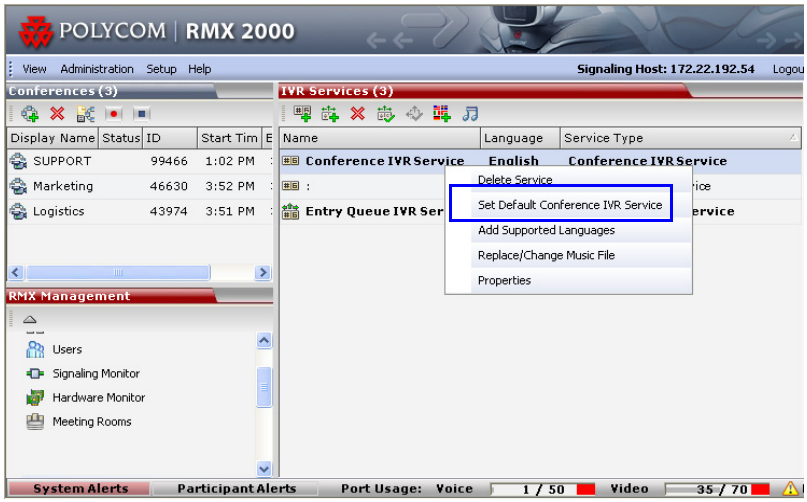
Setting a Conference IVR Service or Entry Queue IVR Service as the Default Service

The first Conference IVR Service and Entry Queue IVR Service are automatically selected by default. The IVR Services (Conference and Entry Queue) shipped with the system are also set as default. If additional Conference IVR Services and Entry Queue IVR Services are defined, you can set another service as the default for each service type.

To select the default Conference IVR Service:

- In the *IVR Services* list, select the Conference IVR Service to be defined as the default, and then click the **Set Default Conference IVR Service** ( ) button.

  Alternatively, in the *IVR Services* list, right-click the Conference IVR Service and then select *Set Default Conference IVR Service*.

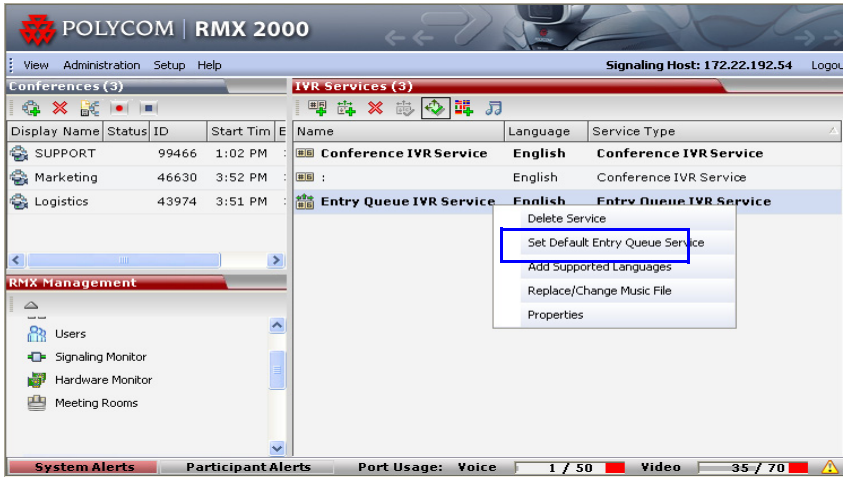


  The IVR Service appears in bold, indicating that it is the current default service.

To select the Default Entry Queue IVR Service:

- In the *IVR Services* list, select the Entry Queue IVR Service to be defined as the default, and then click **Set Default Entry Queue IVR Service** ( ) button.

Alternatively, in the *Conference IVR Services* list, right-click the Entry Queue IVR Service and then select *Set Default Entry Queue IVR Service*.



The default Entry Queue IVR Service appears in bold, indicating that it is the current default service.

Modifying the Conference or Entry Queue IVR Service Properties

You can modify the properties of an existing IVR Service, except the service name and language.

To modify the properties of an IVR Service:

1. In the *RMX Management* pane, click **IVR Services**.
2. In the *IVR Services* list, Click the IVR Service to modify.
   For more information about the tabs and options of this dialog box, see "*Defining a New Conference IVR Service*" on page **13-9**.
3. Modify the required parameters or upload the required audio files.
4. Click **OK**.

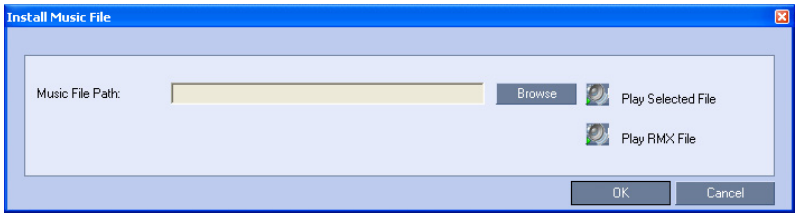
Replacing the Music File

The RMX is shipped with a default music file that is played when participants are placed on hold. For example, while waiting for the chairperson to connect to the conference (if the conference requires a chairperson). You can replace the default music file with your own recorded music. The files include both default IVR and Entry Queue Services. The file must be in *.wav format and its length cannot exceed one hour.

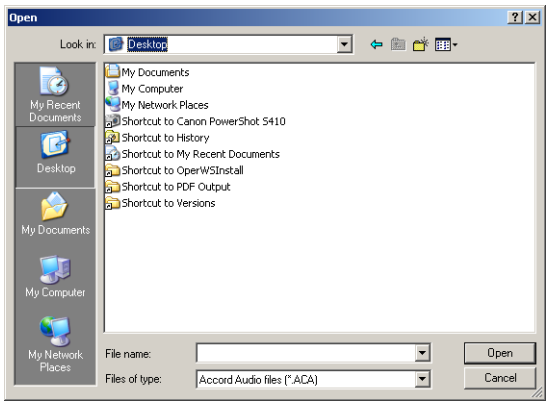
Adding a Music File

To replace the Music file:

1. In the *RMX Management* pane, click **IVR Services**.
2. In the *IVR Services* list toolbar, click the **Replace/Change Music File** ( ) button.
   The *Install Music File* window opens.



3. Click the **Browse** button to select the audio file (*.wav) to upload.
   The *Open* dialog box opens.



4 Select the appropriate audio *.wav file and then click the **Open** button.
The selected file name appears in the *Install Music File* dialog box.

5 Optional. You can play the selected file by clicking the *Play* ( ) button.

   a Click **Play Selected File** to play a file on your computer

   b Click **Play RMX File** to play a file already uploaded on the RMX

6 In the *Install Music File* dialog box, click **OK** to upload the file to the MCU.
The new file replaces the previously uploaded file and this file is used for all background music played by the MCU.

Creating Audio Prompts and Video Slides

The RMX is shipped with default voice messages (in WAV format) and video slides that are used for the default IVR services. You can create your own video slides and record the voice messages for different languages or customize them to your needs.

Recording an Audio Message

To record audio messages, use any sound recording utility available in your computer or record them professionally in a recording studio. Make sure that recorded message can be saved as a Wave file (*.wav format) and that the recorded format settings are as defined in steps 4 and 5 on page 13-36. The files are converted into the RMX internal format during the upload process.
This section describes the use of the Sound Recorder utility delivered with Windows 95/98/2000/XP.

To define the format settings for audio messages:

The format settings for audio messages need to be set only once. The settings will then be applied to any new audio messages recorded.

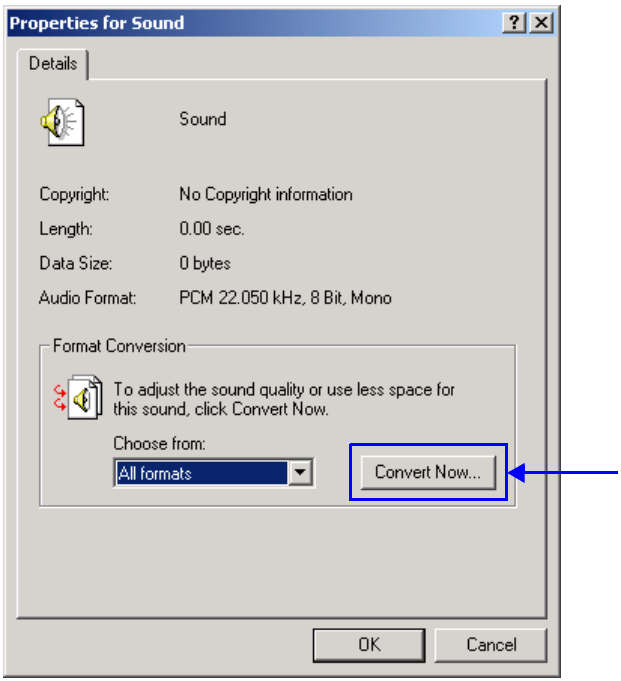
1 On your PC, click **Start > Programs > Accessories > Entertainment > Sound Recorder**.

The *Sound–Sound Recorder* dialog box opens.

2 To define the recording format, click **File > Properties**.

The *Properties for Sound* dialog box opens.

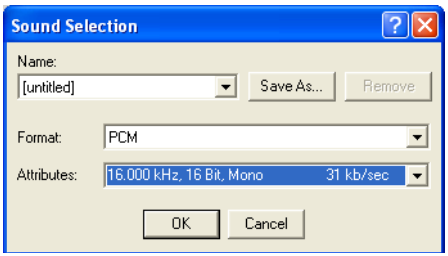
3 Click the **Convert Now** button.



The *Sound Selection* dialog box opens.

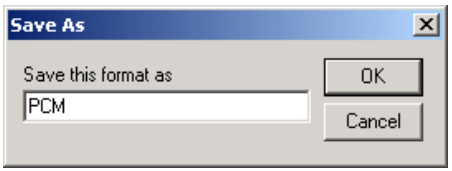
4 In the *Format* field, select **PCM**.

5 In the *Attributes* list, select **16.000 kHz, 16Bit, Mono.**



6 To save this format, click the **Save As** button.
The *Save As* dialog box opens.

7 Select the location where the format will reside, enter a name and then click **OK**.



The system returns to the *Sound Selection* dialog box.

8 Click **OK**.
The system returns to the *Properties for Sound* dialog box.

9 Click **OK**.
The system returns to the *Sound–Sound Recorder* dialog box. You are now ready to record your voice message.

To record a new audio message:

Regardless of the recording utility you are using, verify that any new audio message recorded adheres to the following format settings: **16.000kHz, 16Bit, Mono**.

Make sure that a microphone or a sound input device is connected to your PC.

1 On your PC, click **Start > Programs > Accessories > Entertainment > Sound Recorder**.

The *Sound–Sound Recorder* dialog box opens.

2 Click **File > New**.

3 Click the **Record** button.
The system starts recording.

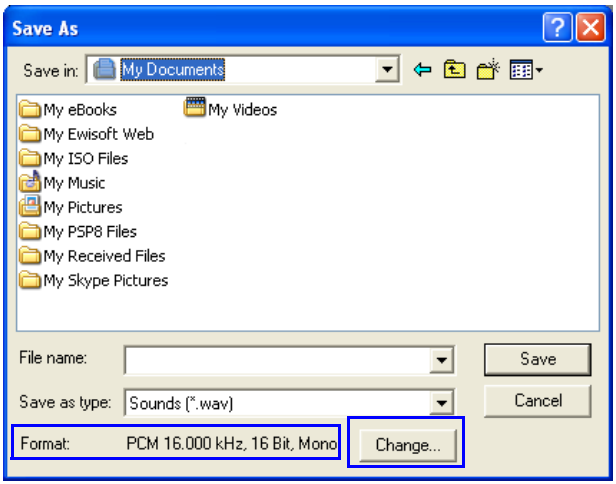
4 Start narrating the desired message.

For all audio IVR messages, stop the recording anytime up to 3 minutes (which is the maximum duration allowed for an IVR voice message). If the message exceeds 3 minutes it will be rejected by the RMX unit.

5 Click the **Stop Recording** button.

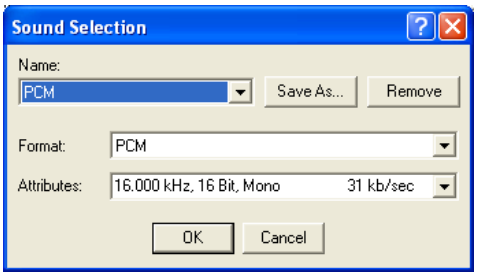
6 Save the recorded message as a wave file, click **File > Save As**.

The *Save As* dialog box opens.



7 Verify that the *Format* reads: **PCM 16.000 kHz, 16Bit, Mono**. If the format is correct, continue with step 10. If the format is incorrect, click the **Change** button.

The *Sound Selection* dialog box appears.



8 In the *Name* field, select the name of the format created in step 7 on page 13-37.

9 Click **OK**.

The system returns to the *Save As* dialog box.

10 In the *Save in* field, select the directory where the file will be stored.

11 In the *Save as Type* field, select the ***.wav** file format.

12 In the *File name* box, type a name for the message file, and then click the **Save** button.

13 To record additional messages, repeat steps 1 to 10.

To upload your recorded *.wav file to the RMX, see step 6 on **page 13-11**.

Creating a Welcome Video Slide

The video slide is a still picture that can be created in any graphic application.

To create a welcome video slide:

1 Using any graphic application, save your image in either *.jpg or *.bmp file format.

2 For optimum quality, verify that the image's dimensions adhere to the RMX's maximum values: Height:1200, Width:1600 pixels.

3 Save your file.

To upload your video slide to the RMX, see step 10 on **page 13-29**.

If using a default Polycom slide, the slide's resolution will be as defined in the profile, i.e. SD, HD or CIF. If using a custom slide, the resolution will be CIF.

Default IVR Prompts and Messages

The system is shipped with the following audio prompts and messages:

Table 13-12 Default IVR Messages

Message Type	Message Text	File Name
General Welcome Message	"Welcome to unified conferencing."	General_ Welcome.wav
Chairperson Identifier Request	"For conference Chairperson Services, Press the Pound Key. All other participants please wait..."	Chairperson_ Identifier.wav
Request Chairperson Password	"Please enter the Conference Chairperson Password. Press the pound key when complete."	Chairperson_ Password.wav
Retry Chairperson Password	"Invalid chairperson password. Please try again."	Chairperson_ Password_ Failure.wav
Request Password	"Please enter the conference password. Press the pound key when complete."	Conference_ Password.wav
Retry Password	"Invalid conference password. Please try again."	Retry_ Conference_ Password.wav
Request Digit	"Press any key to enter the conference."	Request_ Digit.wav
Request Billing Code	"Please enter the Billing code. Press the pound key when complete."	Billing_Code .wav
Requires Chairperson	"Please wait for the chairperson to join the conference."	Requires Chairperson.wav
Chairperson Exit	"The chairperson has left the conference."	Chairperson_ Exit.wav
First to Join	"You are the first person to join the conference."	First to Join.wav

Table 13-12 Default IVR Messages (Continued)

Message Type	Message Text	File Name
Mute All On	"All conference participants are now muted."	Mute_All_On.wav
Mute All Off	"All conference participants are now unmuted."	Mute_All_Off.wav
End Time Alert	"The conference is about to end."	End_Time_Alert.wav
Change Password Menu	"Press one to change conference password. Press two to change chairperson password. Press nine to exit the menu."	Change_Password_ Menu.wav
Change Conference Password	"Please enter the new conference password. Press the pound key when complete."	Change_ Conference_ Password.wav
Change Chairperson Password	"Please enter the new chairperson password. Press the pound key when complete."	Change_ Chairperson_ Password.wav
Confirm Password Change	"Please re-enter the new password. Press the pound key when complete."	Confirm_ Password_ Change.wav
Change Password Failure	"The new password is invalid."	Change_ Password_ Failure.wav
Password Changed Successfully	"The password has been successfully changed."	Password_ Changed_ Successfully.wav
Self Mute	"You are now muted."	Self_Mute.wav
Self Unmute	"You are no longer muted."	Self_Unmute.wav

Table 13-12 Default IVR Messages (Continued)

Message Type	Message Text	File Name
Chairperson Help Menu	"The available touch-tone keypad actions are as follows: • To exit this menu press any key. • To request private assistance, press star, zero. • To request operator's assistance for the conference, press zero, zero. • To mute your line, press star, six. • To unmute your line, press pound, six."	Chairperson_ Help_Menu.wav
Participant Help Menu	"The available touch-tone keypad actions are as follows: • To exit this menu press any key. • To request private assistance, press star, zero. • To mute your line, press star, six. • To unmute your line, press pound, six. • To increase your volume, press star, nine. • To decrease your volume, press pound, nine. • To ask a question, press star, two, two. • To cancel your question, press pound, two, two."	Participant_Help_ Menu.wav
Maximum Participants Exceeded	"The conference is full. You cannot join at this time."	*Maximum_ Participants_ Exceeded*.wav
Roll Call Record	"After the tone, please state your name."	Roll_Call_Record .wav

Table 13-12 Default IVR Messages (Continued)

Message Type	Message Text	File Name
Roll Call Joined	"...has joined the conference."	Roll_Call _Joined .wav
Roll Call Left	"...has left the conference."	Roll_Call_Left .wav
Roll Call Review	"The conference participants are..."	Roll_Call_ Review.wav
Request Conference NID	"Please enter your conference NID. Press the pound key when complete."	Request_ Conference_NID .wav
Retry Conference NID	"Invalid conference NID. Please try again."	Retry_Conference_ NID.wav
Secured Conference	"The conference is now secured."	*Conference_Secured .wav*
Secured Conference	"The conference is now in an unsecured mode"	*Conference_ Unsecured.wav*
Secured Conference	"Conference you are trying to join is locked"	*Conference_Locked. wav*
Conference Recording	"The conference is being recorded"	*Recording_ in_Progress.wav*
Conference Recording	"The conference recording has failed"	Recording_ Failed.wav

Volume Control of IVR Messages, Music and Roll Call

The volume of IVR music, IVR messages and Roll Call is controlled by the following system flags:

- IVR_MUSIC_VOLUME
- IVR_MESSAGE_VOLUME
- IVR_ROLL_CALL_VOLUME

To control the volume of IVR music, messages and Roll Call:

- Modify the values of the *System Flags* listed in Table 13-13 by clicking the menu **Setup > System Configuration**.

  If these flags do not appear in the *System Flags* list, they must be manually added.

  For more information see "*Modifying System Flags*" on page **16-10**.

Table 13-13 *System Flags – IVR Volume Control*

Flag	Description
IVR_MUSIC_VOLUME	The volume of the IVR music played when a single participant is connected to the conference varies according to the value of this flag. Possible value range: 0-10 (Default: 5). 0 – disables playing the music 1 – lowest volume 10 – highest volume
IVR_MESSAGE_VOLUME	The volume of IVR messages varies according to the value of this flag. Possible value range: 0-10 (Default: 6). 0 – disables playing the IVR messages 1 – lowest volume 10 – highest volume **Note:** It is not recommended to disable IVR messages by setting the flag value to 0.

Table 13-13 *System Flags – IVR Volume Control (Continued)*

Flag	Description
IVR_ROLL_CALL_VOLUME	The volume of the Roll Call varies according to the value of this flag. Possible value range: 0-10 (Default: 6). 0 – disables playing the Roll Call 1 – lowest volume 10 – highest volume **Note:** It is not recommended to disable the Roll Call by setting the flag value to 0.

The RMX must be restarted for modified flag settings to take effect.

14

The Call Detail Record (CDR) Utility

The Call Detail Record (CDR) utility enables you to view summary information about conferences, and retrieve full conference information and archive it to a file. The file can be used to produce reports or can be exported to external billing programs.

The value of the fields that support Unicode values, such as the info fields, will be stored in the CDR file in UTF8. The application that reads the CDR must support Unicode.

The Polycom RMX can store details of up to 1000 conferences. When this number is exceeded, the system overwrites conferences, starting with the earliest conference. To save the conferences' information, their data must be retrieved and archived. The frequency with which the archiving should be performed depends on the volume of conferences run by the MCU.

Each conference is a separate record in the MCU memory. Each conference is archived as a separate file. Each conference CDR file contains general information about the conference, such as the conference name, ID, start time and duration, as well as information about events occurring during the conference, such as adding a new participant, disconnecting a participant or extending the length of the conference.

The CDR File

CDR File Formats

The conference CDR records can be retrieved and archived in the following two formats:

- Unformatted data – Unformatted CDR files contain multiple records in "raw data" format. The first record in each file contains general conference data. The remaining records contain event data, one record for each event. Each record contains field values separated by commas. This data can be transferred to an external program such as Microsoft Excel© for billing purposes.

  The following is a sample of an unformatted CDR file:

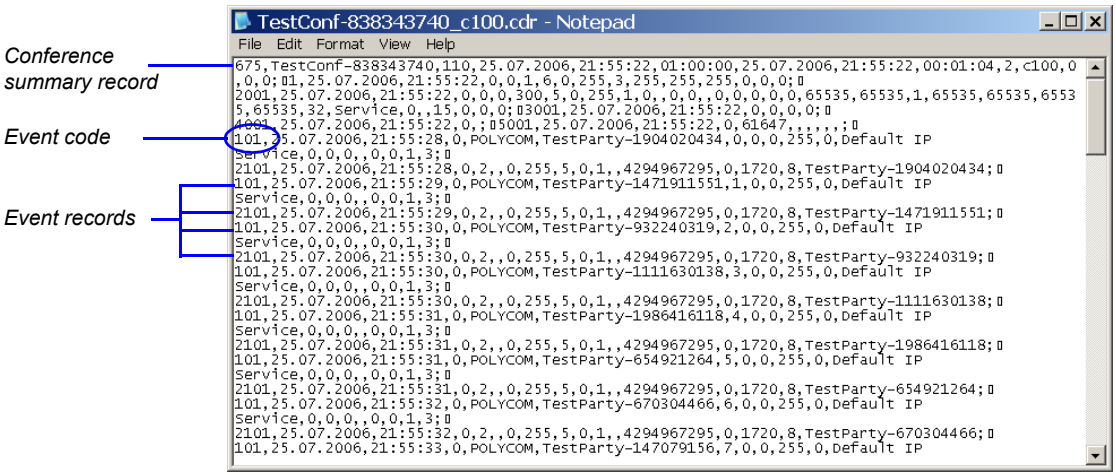


***Figure 14-1** Unformatted CDR File*

- Formatted text – Formatted CDR files contain multiple sections. The first section in each file contains general conference data. The remaining sections contain event data, one section for each event. Each field value appears in a separate line, together with its name. This data can be used to generate a summary report for a conference

The field names and values in the formatted file will appear in the language being used for the *RMX Web Client* user interface at the time when the CDR information is retrieved.

The following is an example of a formatted CDR file:

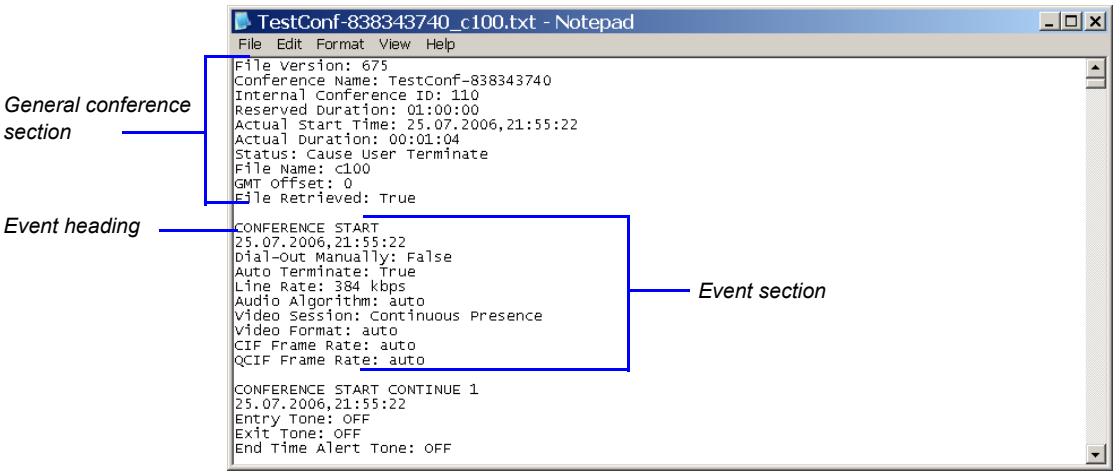


Figure 14-2 Formatted CDR File

CDR File Contents

The general conference section or record contains information such as the Routing Name and ID, and the conference starting date and time.

The event sections or records contain an event type heading or event type code, followed by event data. For example, an event type may be that a participant connects to the conference, and the event data will list the date and time the participant connects to the conference, the participant name and ID, and the participant capabilities used to connect to the conference.

To enable compatibility for applications that written for the MGC family, the Polycom RMX CDR file structure is based on the MGC CDR file structure.

The unformatted and formatted text files contain basically the same information. The following differences should be noted between the contents of the unformatted and formatted text files:

- In many cases a formatted text file field contains a textual value, whereas the equivalent unformatted file field contains a numeric value that represents the textual value.

- For reading clarity, in a few instances, a single field in the unformatted file is converted to multiple fields in the formatted text file, and in other cases, multiple fields in the unformatted file are combined into one field in the formatted file.
- To enable compatibility between MGC CDR files and RMX CDR files, the unformatted file contains fields that were applicable to the MGC MCUs, but are not supported by the RMX MCUs. These fields are omitted from the formatted text file.

Appendix C: "*CDR Fields - Unformatted File"* on page **C-1**, contains a full list of the events, fields and values that appear in the unformatted file. This appendix can be referred to for information regarding the contents of fields in the unformatted text file, but does not reflect the exact contents of the formatted text file.

Viewing, Retrieving and Archiving Conference Information

Viewing the Conference Records

To open the CDR utility:

- On the RMX menu, click **Administration > CDR**.
  The *CDR List* pane opens, displaying a list of the conference CDR records stored in the MCU memory.

CDR List (2000)
Display Name
Start
Duration
Reserved Start Time
Reserved Duration
Status
File Retrieved
Aviv Eisenb 00:02:58 02:00:00 Conference automatically Yes
Default_EQ(00:59:55 01:00:00 Conference terminated w Yes
Aviv Eisenb 00:07:07 02:00:00 Conference automatically Yes
Default_EQ(00:59:55 01:00:00 Conference terminated w Yes
Aviv Eisenb 00:01:17 02:00:00 Conference automatically Yes
Default_EQ(00:59:55 01:00:00 Conference terminated w Yes
Bob Baugh 00:18:01 02:00:00 Conference automatically No
Default_EQ(00:59:55 01:00:00 Conference terminated w No
Holly Dowd 00:01:14 02:00:00 Conference automatically No

The following fields are displayed:

Table 14-1 *Conference Record Fields*

Field	Description
Display Name	The Display Name of the conference and an icon indicating whether or not the CDR record has been retrieved and saved to a formatted text file. The following icons are used: The CDR record has not been saved. The CDR record has been saved.
Start Time	The actual time the conference started.
Duration	The actual conference duration.

Table 14-1 *Conference Record Fields (Continued)*

Field	Description
Reserved Duration	The time the conference was scheduled to last. Discrepancy between the scheduled and the actual duration may indicate that the conference duration was prolonged or shortened.
Status	The conference status. The following values may be displayed: • ***Ongoing Conference*** • ***Terminated by User*** • ***Terminated when end time passed*** • ***Automatically terminated when conference was empty*** – The conference ended automatically because no participants joined the conference for a predefined time period, or all the participants disconnected from the conference and the conference was empty for a predefined time period. • ***Conference never became ongoing due to a problem*** • ***Unknown error*** **Note:** If the conference was terminated by an MCU reset, the status **Ongoing Conference** will be displayed.
File Retrieved	Indicates whether the conference record was retrieved to a formatted text file. (Yes/No)

Refreshing the CDR List

To refresh the CDR list:

- Click the **Refresh** button, or right-click on any record and then select **Refresh**.
  Updated conference CDR records are retrieved from the MCU memory.

Retrieving and Archiving Conference CDR Records

To retrieve and archive CDR records:

1 To retrieve a single CDR record, right-click the record to retrieve and then select the required format (as detailed in Table 14-2). Alternatively, select the record to retrieve, and then click the appropriate button on the toolbar (as detailed in Table 14-2).

To retrieve multiple CDR records simultaneously, use standard Windows multi-selection methods.

Table 14-2 *Conference Information Retrieval Options*

Menu Option	Button	Action
Retrieve		Retrieves the conference information as unformatted data into a file whose extension is .cdr.
Retrieve Formatted XML		Retrieves the conference information as formatted text into a file whose extension is .xml. Note: Viewed when logged in as SUPPORT; SUPPORT
Retrieve Formatted		Retrieves the conference information as formatted text into a file whose extension is .txt.

The *Retrieve* dialog box opens.
The dialog box displays the names of the destination CDR files.

2 Select the destination folder for the CDR files and then click **OK**.

If the destination file already exists, you will be asked if you want to overwrite the file or specify a new name for the destination file.
The files are saved to the selected folder.

15

Gateway Calls

The RMX can be used as a gateway that provides connectivity across different physical networks and translates multiple protocols for point-to-point rich media communications.

The RMX supports the widest range of video and audio algorithms. It allows sites with different frame rates, connection speeds, audio algorithms, video resolutions and network protocols to transparently connect with one another. It also enables multipoint conference creation from an endpoint.

A special conference acting as a *Gateway Session* is created on the RMX. It includes one dial-in connection of the endpoint initiating the *Gateway Session* and one or several dial-out connections to endpoints. It provides connectivity between the various protocols: H.323, SIP, ISDN and PSTN.

To enable the gateway functionality a special Gateway Profile is defined on the RMX.

For details on connectivity between the ISDN/PSTN endpoint and the DMA, see the *RMX 2000 Administrator's Guide*, "*Gateway to Polycom® DMA™ 7000"* on page H-1.

Call Flows

Two calling methods are available:

- Direct (IP participants)
- Via Gateway IVR (IP and ISDN/PSTN participants)

Direct Dialing

This calling method is available to IP participants only and is the recommended method.

The calling endpoint enters the dialing string that includes the access numbers to the RMX Gateway Profile and the number of the destination endpoint. Up to 10 destination numbers can be entered in one string.

The call connects to the RMX *Gateway Profile* and a *Gateway Session* is created. The dial-in participant is automatically connected to it.

During the connection phase, the number being dialed is displayed on the screen of the calling endpoint.

If the call is not answered or it cannot be completed using one communication protocol, the system will try to connect the endpoint using the next communication protocol according to the selected protocols in the following order: H.323, SIP and ISDN. PSTN numbers are identified separately and are dialed immediately without trying other connections.

If the call is busy, the system will not try to connect the endpoint using another protocol.

If the call is not completed after trying all possible protocols, the system displays the number that was dialed on the calling endpoint's screen and the reason for not completing the call. For details, see "*Connection Indications*" on page **1-1**.

When the call is connected, a new *Gateway Session* is created and added to the ongoing *Conferences* list.

Dialing from H.323 Endpoints

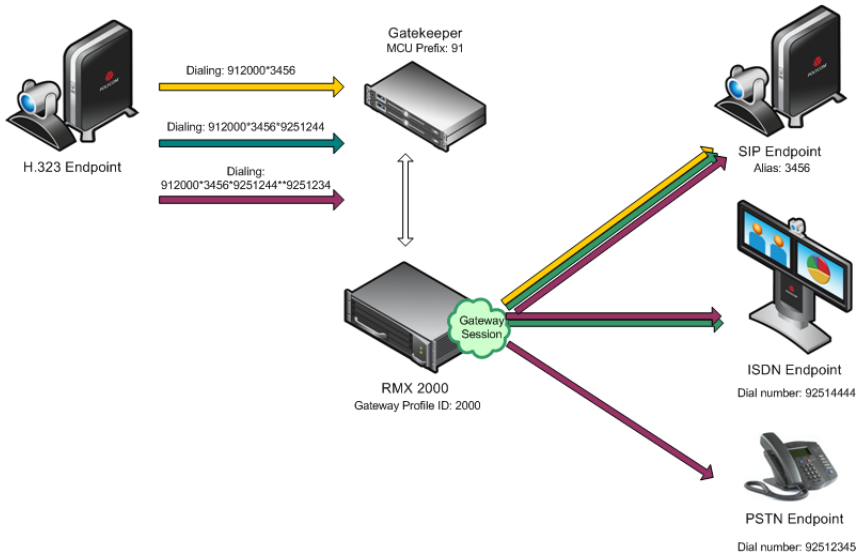


Figure 15-1 Dialing String and Call Flow from H.323 Endpoint to One, Two or Three Endpoints

The calling endpoints can dial to one, two or several endpoints (up to ten) in one dialing string. The dialing string includes the following components:

[MCU prefix in GK] - the prefix with which the RMX is registered to the gatekeeper.

[GW Profile ID] - The ID of the Gateway Profile to be used for routing the call to the destination endpoint or DMA, as defined in the RMX Gateway Profiles. It includes the parameters of the call to the destination.

* - to indicate H.323, SIP or ISDN connection protocol to the destination endpoint (followed by the appropriate destination number). Placing this delimiter before the destination number causes the system to try to connect the endpoint using H.323 first, then SIP and lastly ISDN according to the selected protocols.

** - to indicate a PSTN connection to the destination endpoint (followed by the appropriate destination number).

[Destination number] - the destination number as alias, IPv4 address or ISDN/PSTN number.

The dialing string:

```
[MCU prefix in GK][GW Profile ID]*[Destination Number,
first participant]*[Destination Number, second
participant]**[Destination number].....*[Destination
Number, tenth participant]
```

For example, If the *MCU Prefix in the GK* is 91 and the *GW Profile ID* is 2000, and the destination number is 3456 (SIP) enter: 912000*3456.

To invite two participants: SIP: 3456 and ISDN: 9251444, enter: 912000*3456*9251444.

To invite two participants: SIP: 3456 and a PSTN participant whose number is 9251234, enter: 912000*3456**9251234.

Dialing from SIP Endpoints

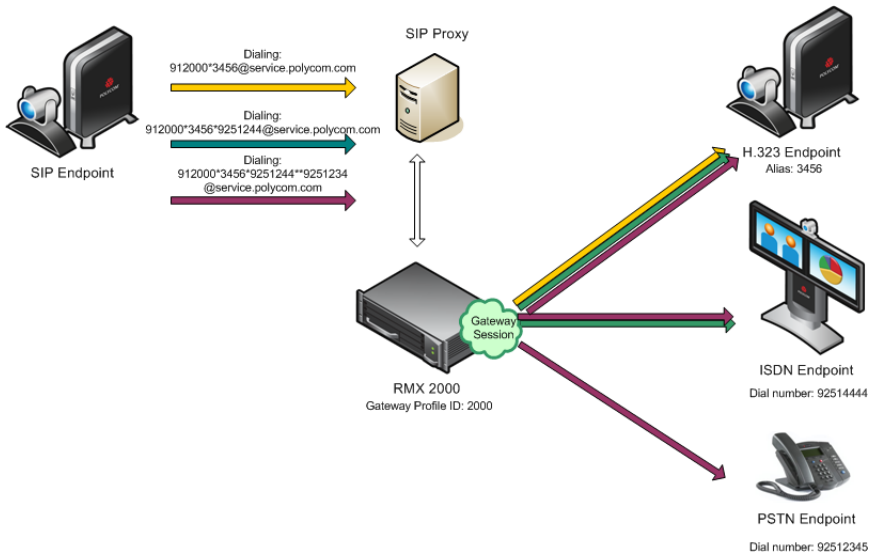


Figure 15-2*Dialing String and Call Flow from SIP Endpoint to One, Two or Three Endpoints*

The calling endpoints can dial to one, two or several endpoints (up to ten) in one dialing string. The dialing string includes the following components:

[MCU Prefix in SIP Proxy] - The prefix with which the RMX is registered to the SIP Proxy. This component is optional and is not required in most cases.

[GW Profile ID] - The ID of the Gateway Profile to be used for routing the call to the destination endpoint or DMA, as defined in the RMX Gateway Profiles. It includes the parameters of the call to the destination.

* - to indicate H.323, SIP or ISDN connection protocol to the destination endpoint (followed by the appropriate destination number). Placing this delimiter before the destination number causes the system to try to connect the endpoint using H.323 first, then SIP and lastly ISDN according to the selected protocols.

** - to indicate a PSTN connection to the destination endpoint (followed by the appropriate destination number).

[Destination number] - the destination number as alias, IPv4 address or ISDN/PSTN number.

[@domain name] - the RMX domain name as registered to the SIP Proxy

The dialing string:

```
[GW Profile ID]*[Destination Number, first
participant]*[Destination Number, second
participant]**[destination number].....*[Destination
Number, tenth participant]@domain name
```

Optional:

```
[GW Profile ID]*[Destination Number, first
participant]*[Destination Number, second
participant]**[destination number].....*[Destination
Number, tenth participant]@IP address of the RMX
signaling host
```

Optional:

```
[MCU prefix in SIP Proxy][GW Profile ID]*[Destination
Number, first participant]*[Destination Number, second
participant]**[destination number].....*[Destination
Number, tenth participant]@domain name
```

For example, if the GW Profile ID is 2000, the domain name is service.polycom.com, and the destination number is 3456, enter: 2000*3456@service.polycom.com.

If using the IP address of the RMX signaling host (for example, 172.22.188.22) instead of the domain name enter: 2000*3456@172.22.188.22.

To invite two participants IP: 3456 and ISDN: 9251444, enter: 2000*3456*9251444@service.polycom.com.

To invite two participants IP: 3456 and PSTN: 9251234, enter: 912000*3456**9251234@service.polycom.com.

Gateway IVR

The calling method for ISDN and PSTN endpoints that can also be used by IP endpoints when the destination dialing string includes the address of the MCU only.

The calling endpoint enters the dialing string that includes the access number to the RMX Gateway Profile.

The endpoint connects to the RMX and is welcomed by the IVR Welcome slide and message: “Please enter the destination number” followed by the dial tone.

Using the endpoint’s DTMF input device such as remote control, the participant enters the number of the destination endpoint followed by the # key. Only one number can be dialed.

While the system dials to the destination endpoints, the participant hears the dialing rings. During the connection phase, the number being dialed is displayed on the screen of the calling endpoint.

If the call is not answered or it cannot be completed using one communication protocol, the system will try to connect the endpoint using the next communication protocol according to the selected protocols in the following order: H.323, SIP and ISDN. PSTN numbers are identified separately and are dialed immediately without trying other connections.

If the endpoint is busy, the system will not try to connect the endpoint using another protocol.

If the call is not completed after trying all possible protocols, the system displays the number that was dialed on the calling endpoint’s screen and the reason for not completing the call. For details, see "*Connection Indications*" on page **1-1**.

Dialing from ISDN/PSTN Endpoints

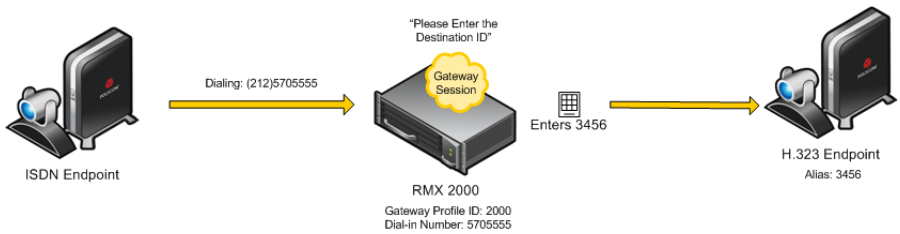


Figure 15-3*Dialing String and Call Flow from ISDN Endpoint to IP Endpoint*

`[GW Profile ISDN/PSTN number]` - the dial-in number assigned to the Gateway Profile, including the required country and area codes.

For example, if the dial-in number assigned to the Gateway Profile is 5705555, enter this number with the appropriate area code: 2125705555.

Once the participant is connected to the *Gateway Profile* and hears the IVR message requesting the destination number, using the DTMF input keypad, the participant enters the number of the destination endpoint followed by the # key. For example, enter 3456# for IP endpoint.

To enter an IP address as the destination number, replace the periods (.) with asterisks (*) in the format `n*n*n*n` followed by the # key. For example, if the IP address is 172.22.188.22, enter 172*22*188*22#.

Dialing from H.323 Endpoints

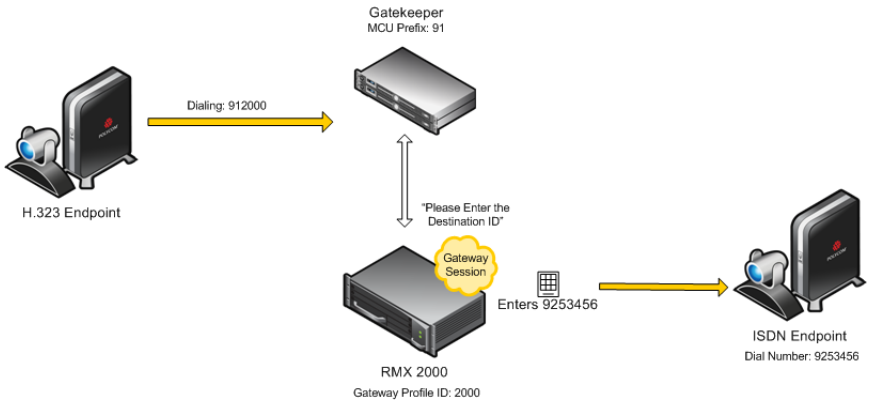


Figure 15-4Dialing String and Call Flow from IP Endpoint to ISDN Endpoint

[MCU prefix in GK] - the prefix with which the RMX is registered to the gatekeeper.

[GW Profile ID] - The ID of the Gateway Profile to be used for the gateway call and the IVR message.

The dialing string format is:

`[MCU prefix in GK][GW Profile ID]`

For example, if the MCU Prefix in the GK is 91 and the GW Profile ID is 2000 enter: 912000.

Once the participant is connected to the *Gateway Profile* and hears the IVR message requesting the destination number, using the DTMF input keypad, the participant enters the number of the destination endpoint followed by the # key. PSTN numbers are identified by an * before the number.

For example, enter 3456# for IP endpoint, or 9253456# for ISDN, or *9253456# for PSTN phone.

To enter an IP address as the destination number, replace the periods (.) with asterisks (*) in the format `n*n*n*n` followed by the # key. For example, if the IP address is 172.22.188.22, enter 172*22*188*22#.

Dialing from SIP Endpoints

Optional. [MCU prefix in SIP Proxy] - the prefix with which the RMX is registered to the gatekeeper.

[GW Profile ID] - The ID of the Gateway Profile to be used for the gateway call and the IVR message.

[@domain name] - the RMX domain name as registered to the SIP Proxy.

The dialing string:

```
[GW Profile ID]@domain name
```

Optional:

```
[GW Profile ID]@IP address of the RMX signaling host
```

Optional:

```
[MCU prefix in SIP proxy][GW Profile ID]@domain name
```

Once the participant is connected to the *Gateway Profile* and hears the IVR message requesting the destination number, using the DTMF input keypad, the participant enters the number of the destination endpoint followed by the # key. PSTN numbers are identified by an * before the number.

For example, enter 3456# for IP endpoint, or 9253456# for ISDN, or *9253456# for PSTN phone.

To enter an IP address as the destination number, replace the periods (.) with asterisks (*) in the format `n*n*n*n` followed by the # key. For example, if the IP address is 172.22.188.22, enter 172*22*188*22#.

Interoperability with CMA

The RMX does not register to the gatekeeper as a Gateway, therefore it is recommended to create and use the CMA *Dialing Rules* to enable the CMA Dial One Method.

When the caller enters the Dial One digit as the destination number prefix, the CMA replaces this digit with the MCU prefix in the Gatekeeper and the ID of the Gateway Profile. For example, the calling participant can enter 99251444, where 9 is the digit that is used as the MCU prefix registered in gatekeeper and is replaced by the gatekeeper with * and the Gateway Profile ID (for example, *2000) as defined in the Dialing Rule.

For more details on Dialing Rules definition in the CMA, see the *Polycom CMA System Operations Guide, "Dial Rule Operations"*.

Connection Indications

During the connection process to the other endpoints, the system displays on the calling participant's screen the called number and the connection status.

A Maximum of 32 characters can be displayed for connection indications. If the displayed information is longer than 32 characters the text is truncated.

If the system dials out to only one destination endpoint, the dialed number is not shown, only the connection status.

If the destination endpoint is ISDN, the system displays the connection progress in percentages, where the percentages represent various stages in the connection process as follows:

- Up to 60% the connection of the ISDN channels (up to 30 channels can be connected when E1 is used for the connection).
- 60% - 80% BONDING stage
- 80% - 90% Capability exchange stage
- 90% - 99% Media connection stage

Once the call is completed, the indications are cleared.

If the call is not completed after trying all possible protocols, the system displays the number that was dialed on the calling endpoint's screen and one of the following causes:

- *Busy* - the far endpoint is in another call. In such a case, the system does not try to connect using another communication protocol.
- *Rejected* - the far endpoint has rejected the call. In such a case, the system will try to connect using another communication protocol.
- *Unreached* - the number could not be resolved by the gatekeeper or the SIP proxy or could not be found on the network. In such a case, the system will try to connect using another communication protocol.
- *Failed* - any reason causing the system not to complete the connection process. In such a case, the system will try to connect using another communication protocol.

You can hide the connection indications by changing the system configuration. For more details, see "*Displaying the Connection Information - System Configuration*" on page **1-1**.

Gateway Functionality

The following features and capabilities are supported in gateway calls:

- *Gateway Sessions* are in CP mode only.
  If High Definition Video Switching is selected in the Profile assigned to the *Gateway Session*, the system ignores this setting and will run the *Gateway Session* in CP mode.
- H.239 Content
- FECC (IP participants)
- Recording. The *Recording Link* is not considered as a participant and therefore, the gateway session will automatically end when only one of the participants remains connected in addition to the recording link. The video of the *Recording Link* is not included in the display of the video of the gateway call.
- Forwarding of DTMF codes from the *Gateway Session* to a conference running on another gateway, MCU or DMA. This enables the participant to enter the required conference and/or chairperson password when connecting to another conference.
  DTMF forwarding is enabled when there are only two participants connected to the *Gateway Session*.
- Up to 80 gateway calls (same as conferences) may be run on a fully configured RMX 2000.
- Gateway Profiles are included in the *Backup* and *Restore Configuration* operations.
- CDR files are generated for *Gateway Sessions* in the same way as for conferences.
- Cascading. To support cascading, the gateway indicates a lower number than the MCU for master-slave relation (directly or through DMA).
- Gateway calls are supported in Microsoft and Avaya environments.

Configuring the Gateway Components on the RMX

To enable gateway calls in the RMX, the following components have to be configured:

- *Conference IVR Service* to be used with the *Conference Profile* assigned to the *Gateway Profile*. The IVR Services are used for *Gateway IVR* connections.
- *Conference Profile* that includes the IVR Service for the Gateway Session and the settings to automatically terminate the Gateway Session: when one participant is still connected or when no participants are connected
- *Gateway Profile* for call routing.

Defining the IVR Service for Gateway Calls

The system is shipped with a default Conference IVR Services for gateway calls named GW IVR Service that enables you to run gateway calls without defining a new Conference IVR Service. This IVR Service includes the following settings:

- *Welcome slide and message* - disabled
- *Conference and Chairperson Passwords* - disabled
- *General Messages* - all messages including the gateway messages and dial tones are selected
- *Roll Call* - disabled
- *Video Services - Click&View* - enabled
- *Video Services - Video Welcome Slide* - **Default_GW_Welcom_Slide**
- *Operator Assistance* - disabled

You can define a new Conference IVR Service to be used for gateway calls. This Conference IVR Service will be assigned to the appropriate Gateway Profile.

To define a new Conference IVR Service for gateway calls:

1. In the *RMX Management* pane, expand the *Rarely Used* list and click the **IVR Services** ( ) entry.
   The list pane displays the *Conference IVR Services* list.
2. On the *IVR Services* toolbar, click the **New Conference IVR Service** ( ) button.
   The *New Conference IVR Service - Global* dialog box opens.

3. In the *Conference IVR Service Name* field, enter a name that will identify this service as a gateway IVR service.
4. Define the IVR Service Global parameters (it is recommended to use the system defaults). For more details, see *RMX 2000 Administrator's Guide*, "*Conference IVR Service Properties - Global Parameters*" on page **13-10**.
5. When defining a gateway IVR Service, the following options should remain disabled:
   - Welcome Messages (in the *Conference IVR Service - Welcome* dialog box).
   - Chairperson Messages (in the *Conference IVR Service - Conference Chairperson* dialog box).
   - Password Messages (in the *Conference IVR Service - Conference Password* dialog box)
6. Click the **General** tab.

   The *General* dialog box lists messages that are played during the conference. These messages are played when participants or the conference chairperson perform various operations or when a change occurs.
7. To assign the appropriate audio file to the message type, click the appropriate table entry, in the *Message File* column. A drop-down list is enabled.
8. From the list, select the audio file to be assigned to the event/ indication.
9. Repeat steps ? and ? to select the audio files for the required messages.

10 For a gateway IVR Service, select the audio file for the following message types:

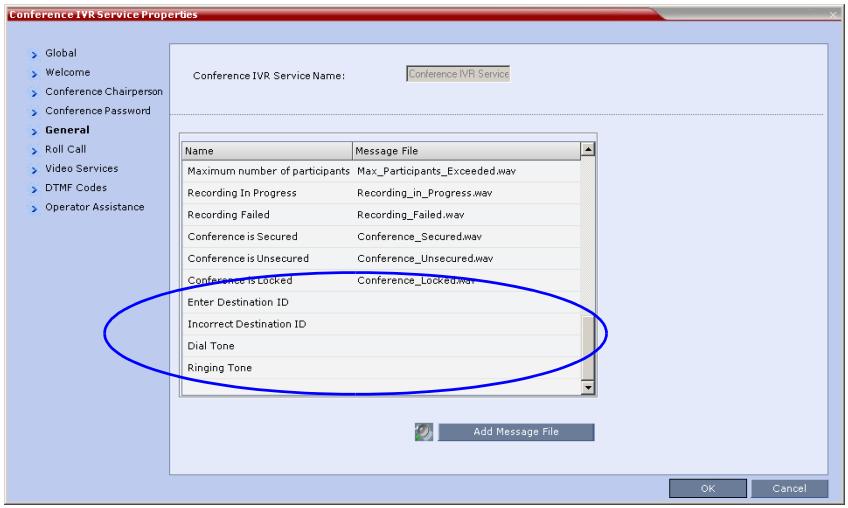


Table 15-1 *Conference IVR Service Properties - Gateway General Voice Messages*

Message Type	Description
Enter Destination ID	Prompts the calling participant for the destination number. Default message prompts the participant for the conference ID (same message as in the Entry Queue IVR Service).
Incorrect Destination ID	If the participant entered an incorrect conference ID (in gateway calls it is the destination number), requests the participant to enter the number again.
Dial Tone	The tone that will be played to indicate a dialing tone, to let the calling participant enter the destination number.
Ringing Tone	The tone that will be played to indicate that the system is calling the destination number.

11 When defining a gateway IVR Service, it is recommended that the *Roll Call* option remains disabled.

12 Click the **Video Services** tab.

The *New Conference IVR Service - Video Services* dialog box opens.

13 Define the following parameters:

Table 15-2 *New Conference IVR Service Properties - Video Services Parameters*

Video Services	Description
Click&View	Select this option to enable endpoints to run the Click&View application that enables participants to select a video layout from their endpoint.
Video Welcome Slide	Select the video slide file to be displayed when participants connect to the conference. To view any slide, click the **Preview Slide** ( ) button. If the video slide file was not uploaded to the MCU prior to the IVR Service definition, click the **Add Slide** button. The *Install File* dialog box opens. The uploading process is similar to the uploading of audio files. For more information, see step 6 on page **13-7** in the *RMX 2000 Administrator's Guide*. **Notes:** • When using one of the default Polycom slides, the slide will be displayed in the resolution defined in the profile, i.e. CIF, SD, HD 720p or HD 1080p. When using a custom slide, it will be displayed in the only in CIF resolution. • When defining a gateway IVR Service, the recommended default slide is: Default_GW_Welcome_Slide.

14 Click the **DTMF Codes** tab.
The *New Conference IVR Service - DTMF Codes* dialog box opens.

15 If required, modify the DTMF codes or permissions. For more details, see *RMX 2000 Administrator's Guide*, *"New Conference IVR Service Properties - DTMF Codes"* on page **13-24**.

16 Click the **Operator Assistance** tab.

17 If Operator Assistance will not be available to participants, clear the **Enable Operator Assistance** option, which is automatically selected to disable it.

18 Click **OK** to complete the IVR Service definition.
The new Conference IVR Service is added to the *IVR Services* list.

Defining the Conference Profile for Gateway Calls

The Conference Profile that will be later assigned to the Gateway Profile determine the parameters of the gateway call, such as the line rate and video resolution and if to automatically terminate the gateway session when one participant or no participants are connected to the *Gateway Session*.

To define a Conference Profile for Gateway Sessions:

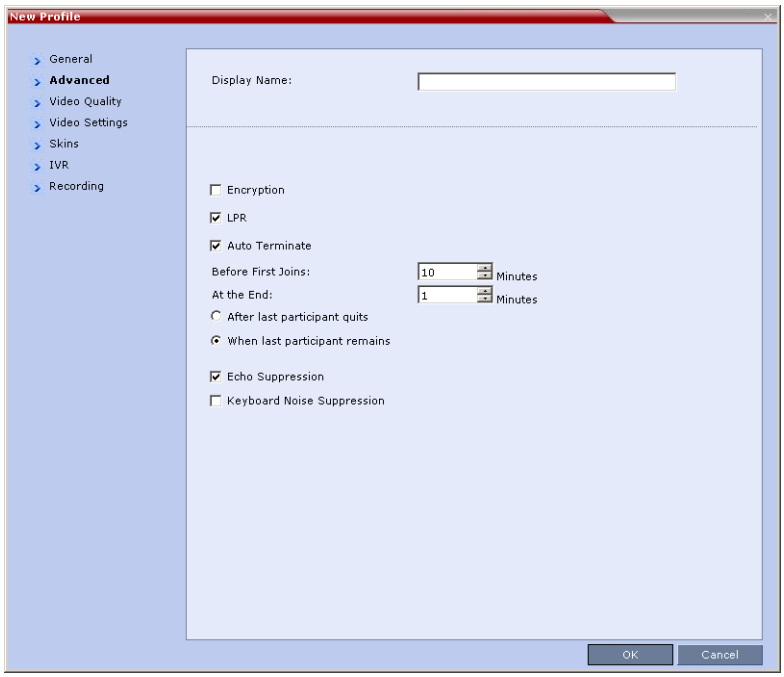
1 In the *RMX Management* pane, click **Conference Profiles**.

2 In the *Conference Profiles* pane, click the **New Profile** button.
The *New Profile – General* dialog box opens.

3 Define the Profile name and select the line rate for the gateway session.

4 Click the **Advanced** tab.

The *New Profile – Advanced* dialog box opens.



5 Define the required settings for Encryption and LPR.

6 Set the *Auto Terminate - At the End* option to **When Last Participant Remains** ensuring that the gateway call will end when only one participant is connected. For more details, see Table 1-5, "New Profile - Advanced Parameters," on page 1-11.

7 Define the remaining Profile parameters as described in "*Defining Profiles*" on page **1-8**.

Defining the Gateway Profile

A Gateway Profile is a conferencing entity, based on the Conference Profile assigned to it, that enables endpoints to dial-in and initiate *Gateway Sessions*. The system is shipped with a default Gateway Profile, named *Default_GW_Session*.

When an endpoint calls the Gateway Profile, a new *Gateway Session* is automatically created based on the Profile parameters, and the endpoint joins the gateway call which can also be a multipoint conference if more than two participants are connected to the conference.

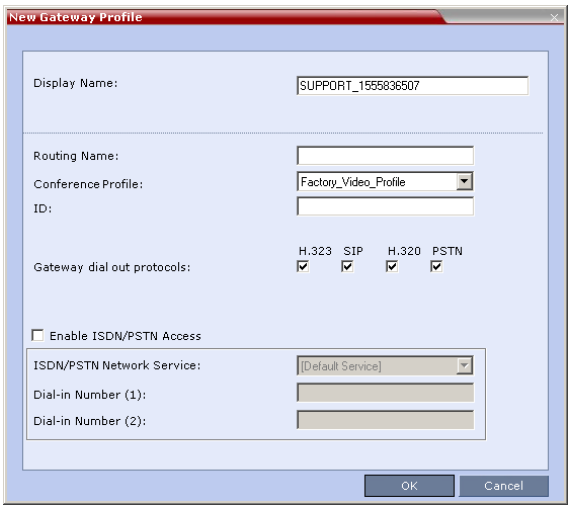
The *Gateway Profile* defines the parameters of the gateway call that are taken from the Conference Profile assigned to it, such as line rate, resolution, the IVR Service to be used and the dial-in numbers.

Up to 1000 Gateway Profiles, Entry Queues, IP Factories and Meeting Rooms can be defined in the RMX (they are all part of one repository whose size is 1000 entries).

To define a new Gateway Profile:

1 In the *RMX Management - Rarely Used* pane, click **Gateway Profiles** .

2 In the *Gateway Profiles* list pane, click the **New Gateway Profile** button.
The *New Gateway Profile* dialog box opens.



3 Define the following parameters:

Table 15-3 *New Gateway Profile Properties*

Option	Description
Display Name	Enter a unique-per-MCU name for the Gateway Profile in native language character sets to be displayed in the RMX Web Client. The system automatically generates an ASCII name for the *Display Name* field that can be modified using Unicode encoding. • English text uses ASCII encoding and can contain the most characters (Maximum length in ASCII is 80 characters). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. The maximum length also varies according to the mixture of Unicode and ASCII.

Table 15-3 *New Gateway Profile Properties (Continued)*

Option	Description
Routing Name	The *Routing Name* is defined by the user, however if no *Routing Name* is entered, the system will automatically assign a new name when the Profile is saved as follows: • If an all ASCII text is entered in *Display Name*, it is used also as the *Routing Name.* • If any combination of Unicode and ASCII text (or full Unicode text) is entered in *Display Name*, the *ID* (such as Conference ID) is used as the *Routing Name.*
Conference Profile	The default Conference Profile is selected by default. If required, select the appropriate Profile from the list of Profiles defined in the MCU. **Note:** In the *Conference Profile - Advance* dialog box, the **Auto Terminate** option enables you to automatically terminate the *Gateway Session* when one participant remains connected (excluding the Recording Link). A new *Gateway Session* is created using the parameters defined in the Profile.
ID	Enter a unique number identifying this conferencing entity for dial in. Default string length is 4 digits. If you do not manually assign the ID, the MCU assigns one after the completion of the definition. The ID String Length is defined by the flag NUMERIC_CONF_ID_LEN in the System Configuration.

Table 15-3 *New Gateway Profile Properties (Continued)*

Option	Description
Gateway dial-out Protocol	Select the communication protocols to be used for dialing out to the destination participant(s). The system starts by connecting the participant using the first selected protocol. If the call is not answered or it cannot be completed using one communication protocol, the system will try to connect the endpoint using the next communication protocol in the following order: H.323, SIP and ISDN. PSTN numbers are identified separately and are dialed right away without trying other connections. By default, all protocols (H.323, SIP, ISDN and PSTN) are selected. Clear the protocol that should not be used for connecting the destination endpoint.
Enable ISDN/ PSTN Dial-in	Select this check box to allocate dial-in numbers for ISDN/PSTN connections. To define the first dial-in number using the default ISDN/PSTN Network Service, leave the default selection. When the Entry Queue is saved on the MCU, the dial-in number will be automatically assigned to the Entry Queue. This number is taken from the dial-in numbers range in the default ISDN/ PSTN Network Service. **Note:** Even if ISDN/PSTN is disabled for dial-in, if an ISDN/PSTN Network Service is defined in the system, and ISDN and/or PSTN are enabled for dialed out, the system will used the default ISDN Network Service for dialing out to the target number.
ISDN/PSTN Network Service	The default Network Service is automatically selected. To select a different ISDN/PSTN Network Service in the service list, select the name of the Network Service.
Dial-in Number (1)	Leave this field blank to let the system automatically assign a number from the selected ISDN/PSTN Network Service. To manually define a dial-in number, enter a required number from the dial-in number range defined for the selected Network Service.

Table 15-3 *New Gateway Profile Properties (Continued)*

Option	Description
Dial-in Number (2)	By default, the second dial-in number is not defined. To define a second-dial-in number, enter a required number from the dial-in number range defined for the selected Network Service.

4 Click **OK**.
The new *Gateway Profile* is added to the list.

Displaying the Connection Information - System Configuration

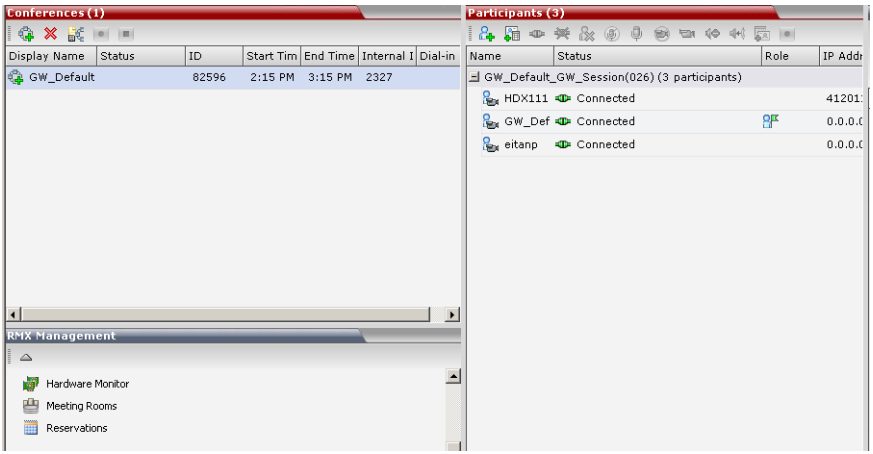
You can hide the connection indications displayed on the participant's screen during the connection phase by changing the system configuration and manually adding and setting the system flag **DISABLE_GW_OVERLAY_INDICATION** to **YES** in the MCMS_PARAMETERS_USER tab.

By default, this flag is set to **NO** and all connection indications are displayed.

For more details about adding and modifying system flags, see *RMX 2000 Administrator's Guide*, *"Manually Adding and Deleting System Flags"* on page **16-19**.

Monitoring Ongoing Gateway Sessions

Ongoing *Gateway Sessions* that are created when calling the Gateway Profile, are listed in the ongoing *Conferences* list pane.



Gateway Sessions are monitored in the same way as the conferences. For more details on monitoring conferences, see *RMX 2000 Administrator's Guide*, "*Conference Level Monitoring*" on page 9-3.

Additional ISDN and PSTN Participants cannot dial in directly to the *Gateway Session* once it was started.

Gateway Session Parameters

Gateway Session Name

The RMX creates a new conference that acts as a *Gateway Session* with a unique ID whose display name is composed of the following components:

- The prefix **GW_**,
- The *Gateway Profile* display name. For example, `Default_GW_Session`
- `(number)` where the number is a gateway conference counter.

For example: if the *Gateway Profile* display name is Default_GW_Session, the conference name will be GW_Default_GW_Session(001).

Conference ID:

The ID of the new conference is assigned randomly by the MCU.

The *Gateway Session* automatically ends when only one participant is left in the session.

Connected Participant Parameters

Once this conference is created, the calling participant is connected to it and one or several dial-out participant(s) are automatically created and added to this *gateway session*. The dial-in participant is also identified as the chairperson of the conference.

The connecting (dial-in) participant name is taken from the endpoint. If the endpoint does not send its name, it is derived from the Gateway Profile display name and it includes the *Gateway Session* name, underscore and a random number (appears between brackets), for example, `GW_Default_GW_Session(001)_(000)`.

The name of the destination (dial-out) participant is taken from the endpoint. If the endpoint does not send its name, it is taken from the dialed number. If the dialed number was an IP address, the system displays underscores instead of dots, for example, `172_22_172_89`.

Participants connected to a *gateway session* are monitored in the same way as participants connected to ongoing conferences. For details, see *RMX 2000 Administrator's Guide*, "*Participant Level Monitoring*" on page **9-14**.

Dialing to Polycom® DMA™ 7000

Audio PSTN/ISDN calls can be routed to Polycom DMA 7000 via the RMX. ISDN Video endpoints connect using their audio channels (but consume video resources). The DMA 7000 enables load balancing and the distribution of multipoint calls on up to 10 Polycom RMX media servers.

As part of this solution, the RMX acts as a gateway for the DMA that supports H.323 calls. The PSTN or ISDN endpoint dials the virtual Meeting Room on the DMA via the Gateway Profile on the RMX.

Both the RMX and the DMA must be registered with the same gatekeeper.

The dialing string of the destination conference on the DMA must be communicated to the dialing endpoint and used during the connection to the Gateway Profile on the RMX.

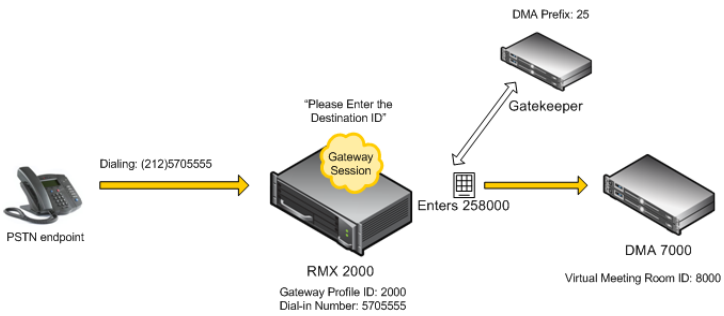


Figure 15-5*Dialing String and Call Flow from ISDN Endpoint to Polycom DMA*

The connection is done in two steps:

- A PSTN/ISDN participant dials the dial-in number assigned to the Gateway Profile (5705555), including the country and area code (if needed) and connects to the Gateway IVR.
- When prompted for the target conference ID, the caller enters the string of the target meeting room on the DMA followed by the # key.

  This string is composed of the DMA prefix as registered in the gatekeeper and the ID of the virtual meeting room running on the DMA. For example, if the DMA prefix is 25 and the target meeting room ID is 8000 the participant enters 258000 followed by the # key.

The RMX creates a *Gateway Session* with two participants, the calling participant and the link to the conference running on the DMA.

Direct Dialing from ISDN/PSTN Endpoint to IP Endpoint via a Meeting Room

Dialing from an ISDN endpoint to a specific IP endpoint using the Gateway Profile is a two-step process (dialing to the Gateway and then entering the number of the destination IP endpoint).

When dialing to specific IP endpoints you can simplify the dialing process by creating the appropriate Meeting Room.

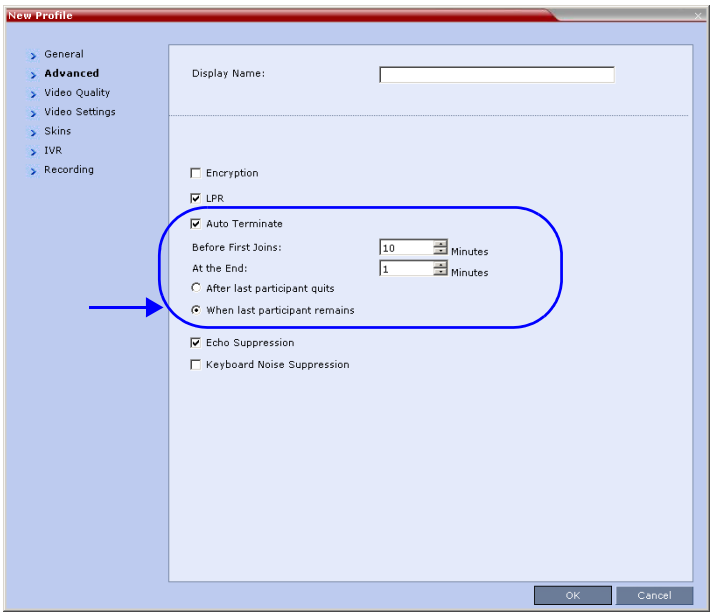
If CMA is involved, dialing can be simplified even further by configuring the appropriate dialing Rule in the CMA.

To set up the Meeting Room for direct dialing in:

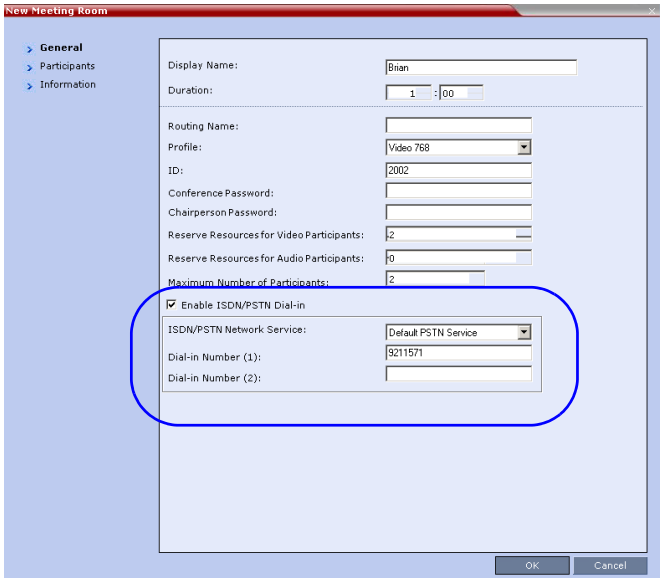
Set the conference parameters in the Conference Profile and make sure that the conference will automatically end when there is only one participant connected to the meeting.

Define the Meeting Room with the following:

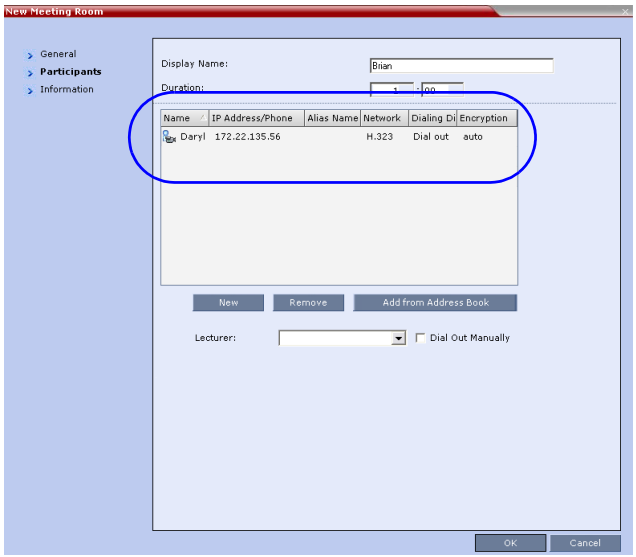
- Conference Profile in which the **Auto Terminate - At the end - When Last Participant Remains** option is selected. For more details on Conference Profile definition, see *"Defining the Conference Profile for Gateway Calls"* on page **15-16**.



- ISDN/PSTN access is enabled and a dial-in number is assigned to the Meeting Room.



- The dial-out IP endpoint is added to the Meeting Room's Participants list.



16

RMX Administration and Utilities

RMX Manager

The *RMX Manager* is the Windows version of the *RMX Web Client*. It can be used instead of the *RMX Web Client* for routine RMX management and for RMX management via a modem connection.

The *RMX Manager* is faster than the *RMX Web Client* and can give added efficiency to RMX management tasks, especially when deployed on workstations affected by:

- Lack of performance due to bandwidth constraints within the LAN/WAN environment.
- Slow operation and disconnections that can be caused by the anti-phishing component of various antivirus applications.

For more information on using the RMX Manager via a modem connection, see Appendix G, "*Connecting to the RMX via Modem*" .

Installing RMX Manager

To install RMX Manager:

The RMX Software must be installed and licensing procedures must be completed before starting the following procedure.

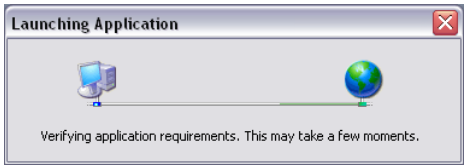
1 Start the *RMX Web Client*.

The Login screen is displayed. There is a link to the *RMX Manager Installer* at the top of the right edge of the screen.

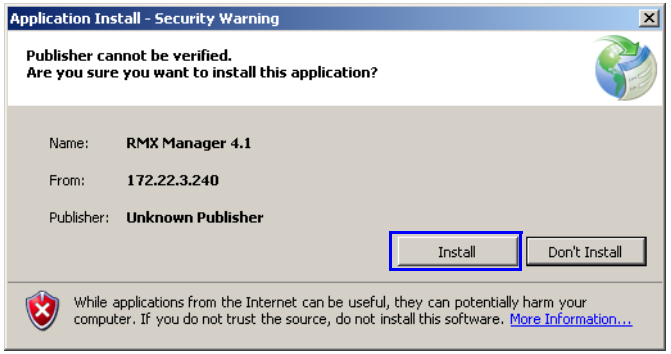
Link to RMX Manager Installer

2 Click the **Install RMX Manager** link.

The installer verifies the application's requirements on the workstation.

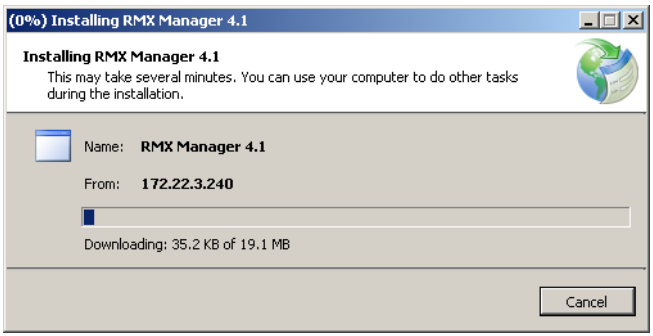


The *Install* dialog box is displayed.

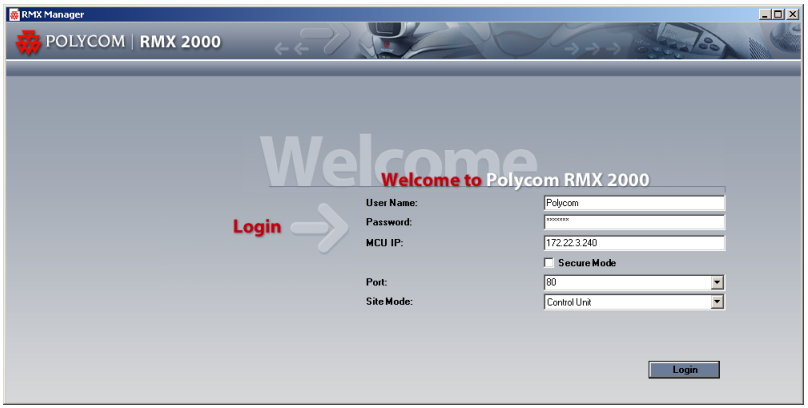

Application Install - Security Warning
Publisher cannot be verified.
Are you sure you want to install this application?
Name: RMX Manager 4.1
From: 172.22.3.240
Publisher: Unknown Publisher
Install
Don't Install
While applications from the Internet can be useful, they can potentially harm your computer. If you do not trust the source, do not install this software. More Information...

3 Click **Install**.

The installation proceeds.


(0%) Installing RMX Manager 4.1
Installing RMX Manager 4.1
This may take several minutes. You can use your computer to do other tasks during the installation.
Name: RMX Manager 4.1
From: 172.22.3.240
Downloading: 35.2 KB of 19.1 MB
Cancel

The installation completes, the application loads and the *RMX Manager – Welcome* screen is displayed with the following information:



— The *MCU IP* address field contains the IP address of the MCU's *Control Unit*.

— The *Port* field contains port number *80*.

— The *Site Mode* field is set to *Control Unit*.

4 In the *Username* field, enter your user name.

5 In the *Password* field, enter your password.

6 **Optional.** To connect with *SSL* and work in *Secure Mode,* select the **Secure Mode** check box.

The *Port* field is automatically set to *443*.

7 **Optional.** To use a different port for *Secure Mode*, select another port from the list or enter the required port number into the *Port* field.

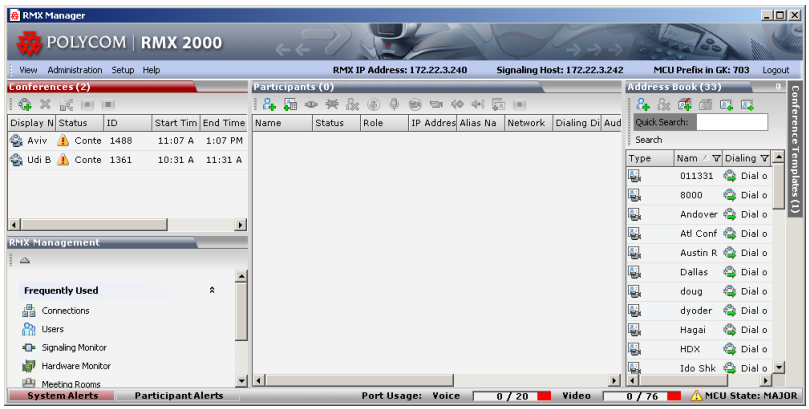
8 **Optional.** To connect to the *Shelf Manager*:

   a In the *Site Mode* field, select **Shelf Management**.

   b In the *MCU IP* field, enter the IP address of the *Shelf Management*.

9 Click **Login**.

   The *RMX Manager* main screen is displayed as a window.



Running RMX Manager

Once installed, the *RMX Manager* can be run using:

— The *http://* command in the browser's address line.

— The Windows *Start* menu.

To use the browser:

- In the browser's command line, enter **`http://<MCU Control Unit IP Address>/RmxManager.html`** and press **Enter**.

To use the Windows Start menu:

1 Click **Start.**

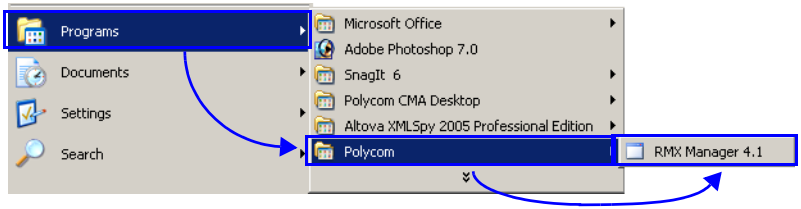
   a If the *RMX Manager* appears in the recently used programs list, click **RMX Manager** in the list to start the application.

   or

   b Click **All Programs.**

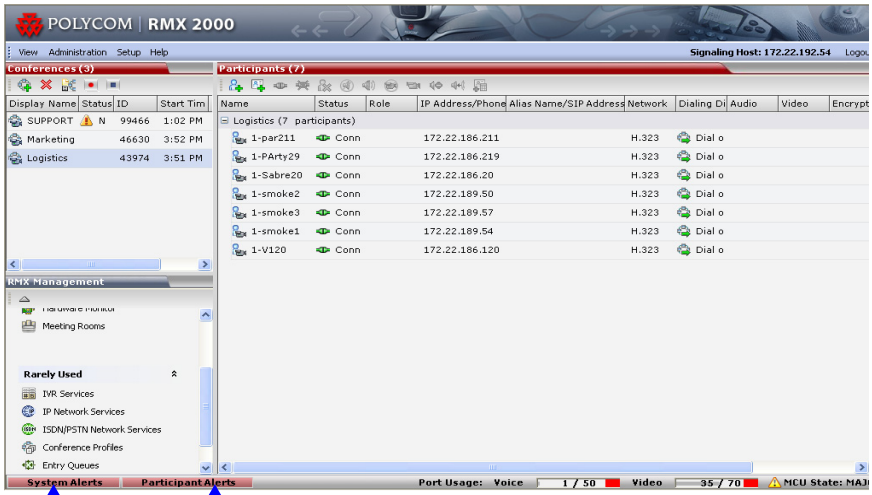
The *All Programs* list is displayed.

2 Select **Polycom** and then select **RMX Manager**.



System and Participant Alerts

The RMX alerts users to any faults or errors the MCU encountered during operation. Two indication bars labeled *System Alerts* and *Participant Alerts* signal users of system errors by blinking red in the event of an alert.


POLYCOM | RMX 2000
View Administration Setup Help
Signaling Host: 172.22.192.54
Conferences (3)
Participants (7)
Display Name Status ID Start Tim
SUPPORT N 99466 1:02 PM
Marketing 46630 3:52 PM
Logistics 43974 3:51 PM
Name Status Role IP Address/Phone Alias Name/SIP Address Network Dialing Di Audio Video Encrypt
Logistics (7 participants)
1-par211 Conn 172.22.186.211 H.323 Dial o
1-PArty29 Conn 172.22.186.219 H.323 Dial o
1-Sabre20 Conn 172.22.186.20 H.323 Dial o
1-smoke2 Conn 172.22.189.50 H.323 Dial o
1-smoke3 Conn 172.22.189.57 H.323 Dial o
1-smoke1 Conn 172.22.189.54 H.323 Dial o
1-V120 Conn 172.22.186.120 H.323 Dial o
RMX Management
Meeting Rooms
Rarely Used
IVR Services
IP Network Services
ISDN/PSTN Network Services
Conference Profiles
Entry Queues
System Alerts
Participant Alerts
Port Usage: Voice 1 / 50 Video 35 / 70
MCU State: MAJ
System Alerts indication bar
Participant Alerts indication bar

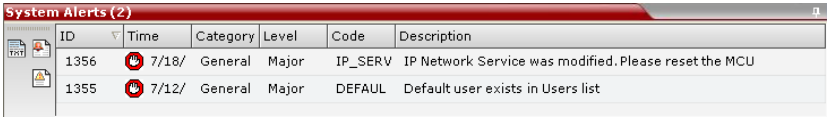
The *System Alerts* indication bar blinks red prompting the user to view the active alarms. Once viewed, the *System Alerts* indication bar becomes statically red until the errors have been resolved in the MCU. The *Participants Alerts* indication bar blinks red indicating participant connection difficulties in conferences. Once viewed, the *Participant Alerts* indication bar becomes statically red until the errors have been resolved in the MCU.

System Alerts

System Alerts are activated when the system encounters errors such as a general or card error. The system errors are recorded by the RMX and can be generated into a report that can be saved in *.txt format.

To view the System Alerts list:

1 Click the red blinking **System Alerts** indication bar.
The *System Alerts* pane opens. This screen indicates what events have not been resolved.



The following columns appear in the *System Alerts* pane:

Table 16-1 *Active Alarms Pane Columns*

Field	Description
ID	An identifying number assigned to the system alert.
Time	Lists the date and time that the error occurred. This column also includes the icon indicating the error level (as listed in the level column).
Category	Lists the type of error. The following categories may be listed: • **File** – indicates a problem in one of the files stored on the MCU's hard disk. • **Card** – indicates problems with a card. • **Exception** – indicates software errors. • **General** – indicates a general error. • **Assert** – indicates internal software errors that are reported by the software program. • **Startup** – indicates errors that occurred during system startup. • **Unit** – indicates problems with a unit. • **MPL** - indicates an error related to a Shelf Management component (MPL component) other than an MPM, RTM or switch board.

Table 16-1 Active Alarms Pane Columns (Continued)

Field	Description
Level	Indicates the severity of the problem, or the type of event. There are three fault level indicators: – Major Error – System Message – Startup Event
Code	Indicates the problem, as indicated by the error category.
Process Name	Lists the type of functional process involved.
Description	When applicable, displays a more detailed explanation of the cause of the problem.

For more information about the Active Alarms, see *Appendix B: "Alarms and Faults"* on page **B-1**.

2 Click one of the following two buttons to view its report in the *System Alerts* pane:

Active Alarms (default) – this is the default reports list that appears when clicking the System Alerts indication bar. It displays the current system errors and is a quick indicator of the MCU status.

Faults List – a list of faults that occurred previously (whether they were solved or not) for support or debugging purposes.

3 To save the *Active Alarms* or *Faults* report to a text file, click the **Save to Text** button.

The *Save* dialog window opens.

4 Select a destination folder and enter the file name.

5 Click **Save**.

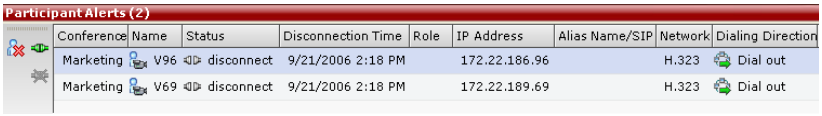
Participant Alerts

Participant Alerts enables users, participants and conferences to be prompted and currently connected. This includes all participants that are disconnected, idle, on standby or waiting for dial-in. Alerts are intended for users or administrators to quickly see all participants that need their attention.

To view the Participants Alerts list:

1 Click the red blinking **Participants Alerts** indication bar.

The *Participant Alerts* pane opens.



The *Participant Alerts* pane displays similar properties to that of the *Participant List* pane. For more information, see the *RMX 2000 Getting Started Guide* - "*Participant Level Monitoring*" on page 3-47.

2 To resolve participants and the alarms they have generated, users can either **Connect** , **Disconnect** or **Delete** a participant.

System Configuration

Aspects of the system's overall behavior can be configured by modifying the default values of system flags.

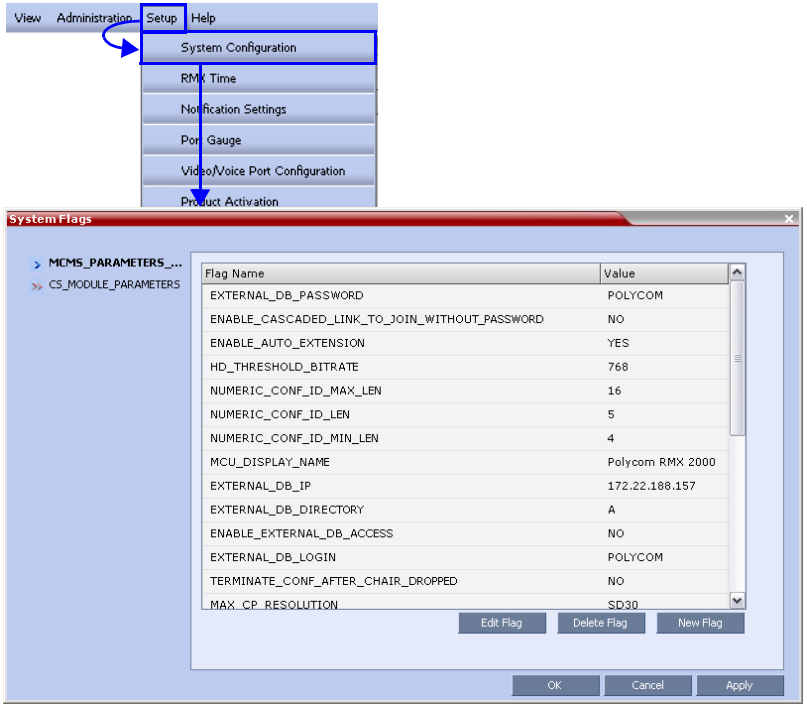
For flag changes to take effect, the MCU must be reset. For more information, see *Chapter 16, "Resetting the RMX"* on page **16-90**.

Modifying System Flags

To modify system flags:

1 On the *RMX* menu, click **Setup > System Configuration**.

The *System Flags* dialog box opens.



2 In the *MCMS_PARAMETERS* tab, the following flags can be modified:

Table 16-2 *System Flags – MCMS_PARAMETERS*

Flag	Description
ALLOW_NON_ ENCRYPT_PARTY_IN_ ENCRYPT_CONF	If YES, allows non-encrypted participants to connect to encrypted conferences. Default: No
AUTHENTICATE_USER	If the external database application is to be used to verify that operators are authorized to log in to the MCU, set the value of this flag to **YES**. If the value of this flag is set to **NO**, the MCU database is used to verify that operators are authorized to log in to the MCU. **Note:** If the flag is set to **YES,** the flow is first to look in the internal DB and then go out to the external one. Flags for SE200 need to be added manually.
*ENABLE_EXTERNAL_* DB_ACCESS	If YES, the RMX connects to an external database application, to validate the participant's right to start a new conference or access a conference. Default: NO
ENABLE _AUTO_ EXTENSION	Allow conferences running on the RMX to be automatically extended as long as there are participants connected. Default: YES
ENABLE_CASCADED_ LINK_TO_JOIN_ WITHOUT_PASSWORD	Enables a cascaded link to enter a conference without a password. Default: NO, for security reasons.
EXTERNAL_ CONTENT_ DIRECTORY	The Web Server folder name. Change this name if you have changed the default names used by the CMA application. Default: /PlcmWebServices

Table 16-2 System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
EXTERNAL_ CONTENT_IP	The IP address of the CMA server. This flag is also the trigger for replacing the internal RMX address book with the CMA global Address Book. When empty, the integration of the CMA address book with the RMX is disabled.
EXTERNAL_ CONTENT_ PASSWORD	The password associated with the user name defined for the RMX in the CMA server.
EXTERNAL_ CONTENT_ USER	The login name defined for the RMX in the CMA server defined in the format: domain name/user name.
EXTERNAL_DB_ DIRECTORY	The URL of the external database application. For the sample script application, the URL is: *<virtual directory>/SubmitQuery.asp*
EXTERNAL_DB_IP	The IP address of the external database server, if one is used. Default: 0.0.0.0
EXTERNAL_DB_LOGIN	The login name defined for the RMX in the external database server. Default: POLYCOM
EXTERNAL_DB_ PASSWORD	The password associated with the user name defined for the RMX on the external database server. Default: POLYCOM
EXTERNAL_DB_PORT	The external database server port used by the RMX to send and receive XML requests/ responses. For secure communications set the value to 443. Default: 5005.

Table 16-2 *System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
H.263_ANNEX_T	Set to NO to send the Content stream without Annex T and enable Aethra and Tandberg endpoints, that do not support Annex T, to process the Content. Default: YES
HD_THRESHOLD_ BITRATE	Sets the minimum bit rate required by endpoints to connect to an HD Conference. Endpoints that cannot support this bit rate are connected as audio only. Range: 384kbps - 4Mbs (Default: 768)
ISDN_COUNTRY_ CODE	The name of the country in which the MCU is located. Default: COUNTRY_NIL
ISDN_IDLE_CODE_E1	The Idle code (silent), transmitted on the ISDN E1 B channels, when there is no transmission on the channels. Default: 0x54
ISDN_IDLE_CODE_T1	The Idle code (silent), transmitted on the ISDN T1 B channels, when there is no transmission on the channels. Default: 0x13
ISDN_NUM_OF DIGITS	When using ISDN Overlap sending dialing mode, this field holds the number of digits to be received by the MCU. Default: 9

Table 16-2 System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
MAX_CP_RESOLUTION	The maximum CP resolution and frame rate that can be supported by the Polycom RMX 2000. Possible flag values: • **HD1080** – High Definition at 30 fps (MPM+) • **HD720** – High Definition at 60 fps (MPM+) • **HD** – High Definition at 30 fps • **SD30** – Standard Definition at 30 fps • **SD15** – Standard Definition at 15 fps • **CIF** – CIF resolution Default: HD1080 For more information see "*Setting the Maximum CP Resolution for Conferencing"* on page **2-5**.
MCU_DISPLAY_NAME	The name of the MCU that is displayed on the endpoint's screen when connecting to the conference. Default: POLYCOM RMX 2000
NUMERIC_CONF_ID_LEN	Defines the number of digits in the Conference ID that will be assigned by the MCU. Enter 0 to disable the automatic assignment of IDs by the MCU and let the Operator manually assign them. Range: 2-16 (Default: 4).
NUMERIC_CONF_ID_MAX_LEN	The maximum number of digits that the user can enter when manually assigning an ID to a conference. Range: 2-16 (Default: 8) **Note:** Selecting 2 limits the number of simultaneous ongoing conferences to 99.

Table 16-2 *System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
NUMERIC_CONF_ID_MIN_LEN	The minimum number of digits that the user must enter when manually assigning an ID to a conference. Range: 2-16 (Default: 4) **Note:** Selecting 2 limits the number of simultaneous ongoing conferences to 99.
TERMINATE_CONF_AFTER_CHAIR_DROPPED	If YES, sets conferences to automatically terminate if the Chairperson disconnects from the conference. Default: YES
MS_ENVIRONMENT	If YES, sets the RMX SIP environment to integrate with Microsoft OCS solution. Default: NO
LEGACY_EP_CONTENT_DEFAULT_LAYOUT	Defines the video layout to be displayed on the screen of the legacy endpoints when switching to Content mode. Default value: **CP_LAYOUT_1P7** (1+7). For a detailed list of possible flag values for the various video layouts, see Table 16-3, "*LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values,"* on page **16-17**.
IVR_ROLL_CALL_USE_TONES_INSTEAD_OF_VOICE	When set to **YES**, the system does not playback the Roll Call names when participants enter or exit the conference. If the voice messages are replaced with tones the system will play these tones instead. The use of tones requires the uploading of the appropriate tone files in *wav format and replacing the *Roll Call Joined* and *Roll Call Left* message files with the tone files in the *Conference IVR Service - Roll Call* dialog box. When the flag is set to **NO**, Roll Call names are announced when participants enter or exit the conference. Default: NO.

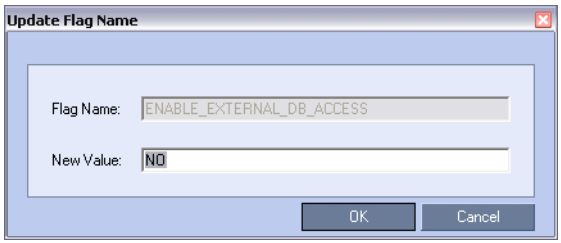
Table 16-2 System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
PAL_NTSC_VIDEO_ OUTPUT	When set to AUTO (default), the video output sent by the RMX is either in PAL or NTSC format. To force the RMX to send the video in either NTSC or PAL, change the flag value accordingly. Default: AUTO.

Currently no flags are defined in the CS_MODULE_PARAMETERS section.

3 To modify a flag value, double-click or select the flag and click the **Edit Flag** button.

4 In the *New Value* field, enter the flag's new value.



5 Click **OK** to close the *Update Flag* dialog box.

6 Repeat steps 2–4 to modify additional flags.

7 Click **OK** to close the *System Flags* dialog box

For flag changes to take effect, reset the MCU. For more information, see the *RMX Administrator's Guide*, "*Resetting the RMX"* on page **16-90**.

LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values

Table 16-3 lists the value for each video layout that can be defined for the LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag. It allows the selection of video layout that will be displayed on the screen of the legacy endpoint when switching to Content mode.

Table 16-3 *LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values*

Layout	Flag Value
	CP_LAYOUT_1X1
	CP_LAYOUT_1X2
	CP_LAYOUT_1X2HOR
	CP_LAYOUT_1X2VER
	CP_LAYOUT_2X1
	CP_LAYOUT_1P2HOR
	CP_LAYOUT_1P2HOR_UP
	CP_LAYOUT_1P2VER
	CP_LAYOUT_2X2
	CP_LAYOUT_1P3HOR_UP
	CP_LAYOUT_1P3VER
	CP_LAYOUT_1P4HOR_UP

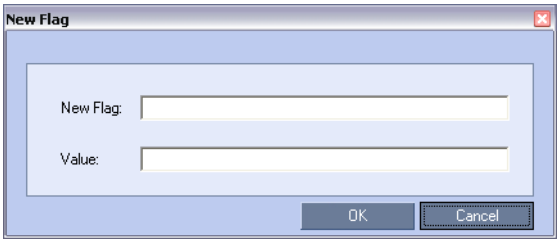
Table 16-3 *LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values (Continued)*

Layout	Flag Value
	CP_LAYOUT_1P4HOR
	CP_LAYOUT_1P4VER
	CP_LAYOUT_1P5
	CP_LAYOUT_1P7
	CP_LAYOUT_1P8UP
	CP_LAYOUT_1P8CENT
	CP_LAYOUT_1P8HOR_UP
	CP_LAYOUT_3X3
	CP_LAYOUT_2P8
	CP_LAYOUT_1P12
	CP_LAYOUT_4X4

Manually Adding and Deleting System Flags

To add a flag:

1 In the *System Flags* dialog box, click the **New Flag** (New Flag) button. The *New Flag* dialog box is displayed.



2 In the *New Flag* field enter the flag name.

3 In the *Value* field enter the flag value.

The following flags can be manually added to the *MCMS_PARAMETERS* tab:

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS*

Flag and Value	Description
SIP_ENABLE_FECC =NO	By default, FECC support for SIP endpoints is enabled at the MCU level. You can disable it by manually adding this flag and setting it to NO.
MIX_LINK _ENVIRONMENT=YES	In H.239-enabled MIH Cascading, when MGC is on level 1, setting this flag to YES will adjust the line rate of HD Video Switching conferences run on the RMX 2000 from 1920Kbps to 17897, 100bits/sec to match the actual rate of the HD Video Switching conference running on the MGC. **Note:** If the flag MIX_LINK_ENVIRONMENT is set to YES, the IP_LINK_ENVIRONMENT flag must be set to NO.

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
IP_ ENVIRONMENT _LINK=NO	In H.239-enabled MIH Cascading, when MGC is on level 1, setting this flag to YES will adjust the line rate of HD Video Switching conferences run on the RMX 2000 from 1920Kbps to 18432, 100bits/sec to match the actual rate of the IP Only HD Video Switching conference running on the MGC. **Note:** If the flag MIX_LINK_ENVIRONMENT is set to NO, the *IP_LINK_ENVIRONMENT* flag must be set to YES.
ENABLE_H239_ANNEX _T=YES	In H.239-enabled MIH Cascading, when MGC is on level 1, enables sending Content using Annex T.
ENABLE_TEXTUAL _CONFERENCE _STATUS=YES	Set the value of this flag to NO to disable *Text Indication.* This setting is recommended for MCUs running Telepresence conferences. Default: YES.
SITE_NAME _TRANSPARENCY =YES	Set the value of this flag to NO to disable *Endpoint Name Transparency.* Default: YES.
SITE_NAMES_ALWAYS _ON =NO	Set the value of this flag to YES to enable the permanent display of *Endpoint Names.* Default: NO.
RMX_MANAGEMENT _SECURITY _PROTOCOL	Enter the protocol to be used for secure communications. Default: TLSV1_SSLV3 (both). Default for U.S. Federal licenses: TLSV1.

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
IVR_MUSIC_VOLUME	The volume of the IVR music played when a single participant is connected to the conference varies according to the value of this flag. Possible value range: 0-10 (Default: 5). 0 – disables playing the music 1 – lowest volume 10 – highest volume
IVR_MESSAGE_VOLUME	The volume of IVR messages varies according to the value of this flag. Possible value range: 0-10 (Default: 6). 0 – disables playing the IVR messages 1 – lowest volume 10 – highest volume **Note:** It is not recommended to disable IVR messages by setting the flag value to 0.
IVR_ROLL_CALL_VOLUME	The volume of the Roll Call varies according to the value of this flag. Possible value range: 0-10 (Default: 6). 0 – disables playing the Roll Call 1 – lowest volume 10 – highest volume **Note:** It is not recommended to disable the Roll Call by setting the flag value to 0.
FORCE_1X1_LAYOUT_ON_CASCADED_LINK_CONNECTION	Set this flag to NO when connecting to an MGC using a cascaded link, if the MGC is functioning as a Gateway and participant layouts on the other network are not to be forced to 1X1. Default: YES

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
ISDN_RESOURCE_POLICY= LOAD_BALANCE (ISDN)	The flag value determines how the ISDN B-channels within configured spans are allocated. The robustness of the ISDN network can be improved by allocating channels evenly (load balancing) among the spans, minimizing the effect of channel loss resulting from the malfunction of a single span. Set the flag value to: • LOAD_BALANCE to allocate channels evenly among all configured spans. • FILL_FROM_FIRST_CONFIGURED_SPAN To allocate all channels on the first configured span before allocating channels on other spans. • FILL_FROM_LAST_CONFIGURED_SPAN To allocate all channels on the last configured span before allocating channels on other spans. Default: LOAD_BALANCE
BONDING_CHANNEL_DELAY (ISDN)	When connecting a bonding group, this is the delay (number of 1/100 seconds) between dialing attempts to connect sequential channels. The channel per second connection performance of ISDN switches can vary and can cause timing issues that result in bonding channel disconnection. Default: 6
BONDING_GROUP_DELAY (ISDN)	When connecting several bonding groups, this is the delay (number of 1/100 seconds) before the first dialing attempt to connect next bonding group. Default: 500

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
BONDING_DIALING _METHOD (ISDN)	When set to: • SEQUENTIAL The MCU initiates channel connections sequentially until it reaches the number of channels defined by the BONDING_NUM_CHANNELS_IN _GROUP flag. When a channel is connected, dialing begins for the next channel in the group. • BY_TIMERS The MCU initiates channel connections sequentially using the values of the BONDING_CHANNEL _DELAY and BONDING_GROUP_DELAY flags. The first group of channels is dialed, using the BONDING_CHANNEL_DELAY between dialing attempts for each channel in the group. The RMX then implements the BONDING_GROUP_DELAY, before dialing the first channel of the next group. Default: SEQUENTIAL
BONDING_NUM_CHANNELS _IN_GROUP (ISDN)	The number of channels in the bonding group to be connected before dialing the next sequential channel. Default: 50
DELAY_BETWEEN _H320_DIAL_OUT _PARTY (ISDN)	The delay in milliseconds that the MCU waits when connecting dial out ISDN and PSTN participants. Default: 1000

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
H323_FREE_VIDEO _RESOURCES	For use in the Avaya Environment. In the Avaya Environment there are features that involve converting undefined dial-in participants' connections from video to audio (or vice versa). To ensure that the participants' video resources remain available for them, and are not released for use by Audio Only calls, set this flag to **NO**. If set to YES, the RMX will release video resources for *Audio Only* calls. Default: YES.
SIP_FREE_VIDEO _RESOURCES	For use in Avaya and Microsoft Environments. When set to **NO** (required for Avaya and Microsoft environments), video resources that were allocated to participants remain allocated to the participants as long as they are connected to the conference even if the call was changed to audio only. The system allocates the resources according to the participant's endpoint capabilities, with a minimum of 1 CIF video resource. Enter YES to enable the system to free the video resources for allocation to other conference participants. The call becomes an audio only call and video resources are not guaranteed to participants if they want to add video again. Default value in Microsoft environment: NO.
CP_REGARD_TO _INCOMING_SETUP _RATE	For use in the Avaya Environment. If set to YES, the RMX calculates the line rate for incoming calls in CP conferences, according to the line rate which is declared by the endpoint in the H.225 setup message. If set to NO, the rate is calculated according to the conference line rate regardless of the rate in the H.225 setup message. Default: YES.

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
H245_TUNNELING	For use in the Avaya Environment. This flag is defined in the *System Flags – CS_MODULE_PARAMETERS* section. In the Avaya Environment, set the flag to YES to ensure that H.245 is tunneled through H.225. Both H.245 and H.225 will use the same signaling port. Default: NO.
H239_FORCE_ CAPABILITIES	When the flag is set to NO, the RMX only verifies that the endpoint supports the Content protocols: Up to H.264 or H.263. When set to YES, the RMX checks frame rate, resolution and all other parameters of the Content mode as declared by an endpoint before receiving or transmitting content. Default: NO.
ENABLE_H239	When set to YES, Content is sent via a separate Content channel. Endpoints that do not support H.239 Content sharing will not be able to receive Content When set to NO, the Content channel is closed. In such a case, H.239 Content is sent via the video channel ("people" video) enabling endpoints that do not support H.239 Content sharing to receive the Content in their video channel. Default: YES.

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
SIP_FAST_UPDATE_ INTERVAL_ENV	Default setting is **0** to prevent the RMX from automatically sending an Intra request to all SIP endpoints. Enter **n** (where n is any number of seconds other than 0) to let the RMX automatically send an Intra request to all SIP endpoints every n seconds. It is recommended to set the flag to 0 and modify the frequency in which the request is sent at the endpoint level (as defined in the next flag).
SIP_FAST_UPDATE_ INTERVAL_EP	Default setting is 6 to let the RMX automatically send an Intra request to Microsoft OC endpoints only, every 6 seconds. Enter any other number of seconds to change the frequency in which the RMX send the Intra request to Microsoft OC endpoints only. Enter 0 to disable this behavior at the endpoint level (not recommended).
FORCE_CIF_PORT_ ALLOCATION	Sets the MCU to allocate one CIF video resource to an endpoint, regardless of the resolution determined by the Conference Profile parameters. You can specify the endpoint types for which resource allocation can be forced to CIF resource, enabling other types of endpoints to use higher resolutions in the same conference. Enter the product type to which the CIF resource should be allocated. Possible values are: • **CMA Desktop -** for CMA desktop client • **VSX nnnn** - where nnnn represents the model number for example, VSX 8000.

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
DISABLE_GW_ OVERLAY_INDICATION	When set to **NO** (default), displays progress indication during the connection phase of a gateway call. Set the value to **YES** to hide the connection indications displayed on the participant's screen during the connection phase of a gateway call.
FORCE_STATIC_MB_ ENCODING	This flag supports Tandberg MXP mode of sending and receiving video by IP endpoint in HD 720p resolution and Video Quality set to Motion. This mode is not supported for ISDN endpoints. Default value: **Tandberg MXP**. To disable this flag, enter **NONE**.
ENABLE_CLOSED_ CAPTION	Enables or disables the Closed Captions option that allow endpoints to endpoints to provide real-time text transcriptions or language translations of the video conference. When set to **NO (default)**, Closed Captions are disabled. When set to **YES**, Closed Captions are enabled.
ENABLE_CISCO_GK	When set to **YES**, it enables the use of an identical prefix for different RMXs when registering with a Cisco MCM Gatekeeper. Default: NO.
SEND_WIDE_RES_TO_ ISDN	When set to **YES**, the RMX sends wide screen resolution to ISDN endpoints. When set to **NO** (default), the RMX does not send wide screen resolution to ISDN endpoints. Default: NO.

Table 16-4 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag and Value	Description
SEND_WIDE_RES_TO_IP	When set to **YES** (default), the RMX sends wide screen resolution to IP endpoints. Endpoint types that do not support wide screen resolutions are automatically identified by the RMX according to their product type and version and will not receive the wide resolution even when the flag is set to YES. When manually added and set to **NO**, the RMX does not send wide screen resolution to all IP endpoints. Default: YES.
DISABLE_WIDE_RES_TO_SIP_DIAL_OUT	When set to **NO** (default), the RMX sends wide screen resolution to dial-out SIP endpoints. Endpoint types that do not support wide screen resolutions are automatically identified by the RMX according to their product type and version and will not receive the wide resolution even if the flag is set to YES. When manually added and set to **YES**, the RMX does not send wide screen. Default: NO.
FORCE_RESOLUTION	Use this flag to specify IP (H.323 and SIP) endpoint types that cannot receive wide screen resolution and that were not automatically identified as such by the RMX. Possible values are endpoint types, each type followed by a semicolon. For example, when disabling Wide screen resolution in an HDX endpoint enter the following string: **HDX;** **Note:** Use this flag when the flag SEND_WIDE_RES_TO_IP is set to YES.

4 Click **OK** to close the *New Flag* dialog box.
The new flag is added to the flags list.

5 Click **OK** to close the *System Flags* dialog box.

For flag changes to take effect, reset the MCU. For more information, see the *RMX Administrator's Guide*, "*Resetting the RMX*" on page **16-90**.

To delete a flag:

1 In the *System Flags* dialog box, select the flag to delete and click the **Delete Flag** button.

2 In the confirmation message box, click **Yes** to confirm.

3 Click **OK to** close the *System Flags* dialog box.

Auto Layout Configuration

The *Auto Layout* option lets the RMX automatically select the conference video layout based on the number of participants currently connected to the conference. You can modify the default selection of the conference video layout to customize it to your conferencing preferences.

Customizing the Default Auto Layout

The default *Auto Layout* is controlled by 11 flags:
`PREDEFINED_AUTO_LAYOUT_0`, ... , **`PREDEFINED_AUTO_LAYOUT_10`**
Each of the 11 *Auto Layout* flags can be left at its default value, or set to any of the *Possible Values* listed in Table 16-5.
The flag that controls the *Auto Layout* you wish to modify must be added to the *System Configuration* file. For more information see "*Modifying System Flags*" on page **16-10**.

Table 16-5 *Flags: PREDEFINED_AUTO_LAYOUT_0,...,10*

Flag Name: **PREDEFINED_AUTO_LAYOUT_n** (n = Number of Participants)		
n	**Default Value**	**Possible Values**
0	CP_LAYOUT_1X1	CP_LAYOUT_1X1
1	CP_LAYOUT_1X1	CP_LAYOUT_1X2
2	CP_LAYOUT_1X1	CP_LAYOUT_1X2HOR
3	CP_LAYOUT_1x2VER	CP_LAYOUT_1X2VER
4	CP_LAYOUT_2X2	CP_LAYOUT_2X1
5	CP_LAYOUT_2X2	CP_LAYOUT_1P2HOR
6	CP_LAYOUT_1P5	CP_LAYOUT_1P2HOR_UP

Table 16-5 *Flags: PREDEFINED_AUTO_LAYOUT_0,...,10 (Continued)*

Flag Name: PREDEFINED_AUTO_LAYOUT_n (n = Number of Participants)		
n	**Default Value**	**Possible Values**
7	CP_LAYOUT_1P5	CP_LAYOUT_1P2VER
8	CP_LAYOUT_1P7	CP_LAYOUT_2X2
9	CP_LAYOUT_1P7	CP_LAYOUT_1P3HOR_UP
10	CP_LAYOUT_1P7	CP_LAYOUT_1P3VER
		CP_LAYOUT_1P4HOR
		CP_LAYOUT_1P4HOR_UP
		CP_LAYOUT_1P4VER
		CP_LAYOUT_1P5
		CP_LAYOUT_1P7
		CP_LAYOUT_1P8UP
		CP_LAYOUT_1P8CENT
		CP_LAYOUT_1P8HOR_UP
		CP_LAYOUT_3X3
		CP_LAYOUT_2P8
		CP_LAYOUT_1P12
		CP_LAYOUT_4X4

Example:

Table 16-6 illustrates the effect of modifying the **PREDEFINED_AUTO_LAYOUT_5** flag in conferences with fewer or more participants than the number of windows selected in the default layout.

Table 16-6 *Example: Modifying PREDEFINED_AUTO_LAYOUT_5 Flag*

<table>
<tr><th>Flag</th><th>Set to Possible Value</th><th>Number of Participants</th><th>Participant's View</th></tr>
<tr><td rowspan="4">PREDEFINED_AUTO_LAYOUT_5

Default =</td><td rowspan="2">CP_LAYOUT_1x2VER</td><td>3</td><td rowspan="2">
Voice activated switching displays the current speaker in the left window of the video layout and only the two last speakers are displayed.</td></tr>
<tr><td>7</td></tr>
<tr><td rowspan="2">CP_LAYOUT_1P5</td><td>3</td><td>
Voice activated switching displays the current speaker in the large (top left) window of the video layout.</td></tr>
<tr><td>7</td><td>
Voice activated switching displays the current speaker in the top left window of the video layout.</td></tr>
</table>

RMX Time

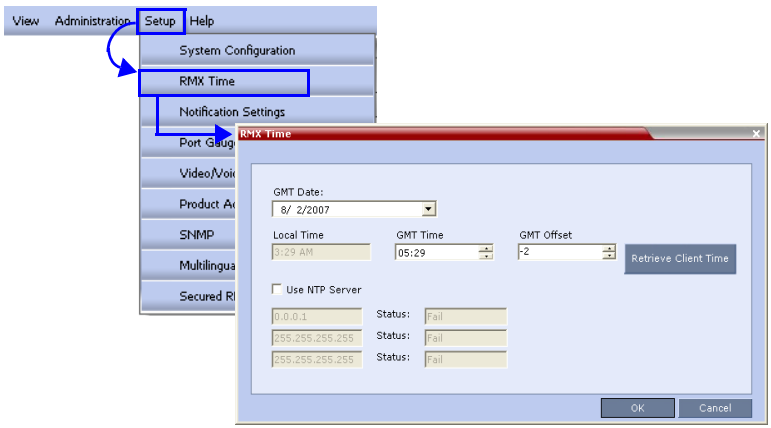
To ensure accurate conference scheduling, the RMX MCU has a clock that can function in standalone mode, or in synchronization with up to three *Network Time Protocol* (NTP) servers.

Altering the clock

The MCU's date and time can be set manually or enabled to synchronize with external NTP servers.

To Alter the RMX Time:

1 On the *RMX* menu, click **Setup > RMX Time** to open the *RMX Time* dialog box.



2 Define/view the following fields:

Table 16-7 RMX Time – Fields Properties

Field	Description
GMT Date	The date at Greenwich, UK.
Local Time	The MCU's local time settings, are based and calculated from the GMT Time and the GMT Offset.

Table 16-7 *RMX Time – Fields Properties (Continued)*

Field	Description
GMT Time	The MCU's current GMT time settings. Select the *Up* or *Down* cursor to alter the *GMT Time* on the MCU.
GMT Offset	The time zone difference between Greenwich and the MCU's location. Select the up or down cursor to alter the *GMT Offset* time on the MCU.
Get Client Time	Click this button to automatically update the MCU's date, time and time zone to match that of the workstation.
Use NTP Server	Select this check box to synchronize the time with up to three NTP servers. When selected, the manual GMT Date & Time setting options are disabled. However, to implement this mode you are required to enable an external connection with an NTP server. Enter the IP addresses of the required NTP servers in order of precedence. The *Status* field indicates whether registration with the NTP Server failed or succeeded.

After resetting the MCU a delay may occur when synchronizing with the external NTP server.

Resource Management

Resource Capacity

The RMX can support two kinds of cards: *MPM* and *MPM+*.

MPM+ cards offer higher capacity additional video capabilities. Three *MPM+* card assemblies are available, *MPM+ 80*, *MPM+ 40* and *MPM+ 20,* offering various port capacities.

Table 16-8 summarizes the resource capacities of an RMX with the various card types.

Table 16-8 *Resource Capacity*

Resource Type	Maximum Possible				
	2 x MPM	MPM+20	MPM+40	MPM+80	2 x MPM+80
Voice	400	100	200	400	800
CIF	80	20	40	80	160
SD30	20	8	15	30	60
HD720p30	20	5	10	20	40
HD720p60	–	2	5	10	20
HD1080p30	–	2	5	10	20

- RMX's with 500MB of memory can support a maximum of 400 simultaneous participant calls, regardless of how system resources are allocated. This limitation applies to RMX's configured with either MPM or MPM+ cards. RMX's with 1000MB of memory are not subject to this limitation.
- RMX memory size is listed in the *Administration* > *System Information* properties box. For more information see "*System Information"* on page **16-56**.

Resource Capacity Modes

The installed media card type (*MPM* and *MPM+)* determines the *Card Configuration Mode*, which in turn determines the resource allocation method that can be selected for the RMX. It determines how the system resources are allocated to the connecting endpoints.

The *System Card Configuration Mode* determines the resource allocation method used by the RMX to allocate resources to the connecting endpoints. The method in which the system allocates the resources is defined in the *Video/Voice Port Configuration*. Two allocation methods are available:

- **Flexible Resource Capacity™ –** This is the same as the allocation mode used in all previous versions. The system allocates the resources according to the connecting endpoints. This mode offers flexibility in resource allocation and is available in both *MPM* and *MPM+ Card Configuration Modes*.

  In *Flexible Resource Capacity* mode, in both *MPM* and *MPM+ Card Configuration Modes,* the maximum number of resources is based on the system license, regardless of the hardware configuration of the RMX. These resources are allocated as CIF resources by default.

  Example: If the RMX is licensed for 80 video resources, but only one *MPM* card is currently installed in the RMX, the system lets you allocate 80 ports although only 40 video resources are available for participant connection. (However, an active alarm will be added to the *Active Alarms* list indicating a resource deficiency).

- **Fixed Resource Capacity™ –** This mode offers precise usage of resources, allowing the administrator to set the number of resources guaranteed to each *Audio Only* and video connection type in advance. This mode is available only in *MPM+ Card Configuration Mode*.

  In *Fixed Resource Capacity* mode, the maximum number of resources is based on the system license and the hardware configuration of the RMX. By default, these resources are allocated as HD720p30 resources, the first time *Fixed Resource Capacity* mode is activated.

  Example: If two *MPM+* cards are installed in the RMX, providing 160 video resources, and the license was not upgraded accordingly, although the system capacity is higher, resource availability for allocation does not change and remains according to the license (80). Conversely, if two *MPM+* cards are installed in the RMX, providing

160 video resources, and the license is for 160 video resources, and one of the *MPM+* cards is removed, the resource availability for allocation is changed to 80.

Resource Usage

Continuous Presence

Video resources usage varies according to the video resolution used by the endpoints. The higher the video resolution (quality), the greater the amount of video resources consumed by the MCU.

Table 16-9 shows the number of video resources used for each resolution on a fully licensed *RMX 2000* configured with either two *MPM* cards or two *MPM+* cards.

Table 16-9 *Video Resource Usage vs. Resolution (MPM, MPM+)*

Resolution/fps	Video Resources Used	
	MPM	MPM+
CIF/30	1	1
QCIF/30		
SIF/30		
WCIF/25	2	2.66
WSIF/30		
432X336/30		
480X352/30		
4CIF/15		
SD/15		
WSD/15		

Table 16-9 *Video Resource Usage vs. Resolution (MPM, MPM+) (Continued)*

<table>
<tr><th rowspan="2">Resolution/fps</th><th colspan="2">Video Resources Used</th></tr>
<tr><th>MPM</th><th>MPM+</th></tr>
<tr><td>WSD/30</td><td rowspan="8">4</td><td rowspan="8">2.66</td></tr>
<tr><td>4CIF/30</td></tr>
<tr><td>4SIF/30</td></tr>
<tr><td>WVGA/30</td></tr>
<tr><td>WVGA/25</td></tr>
<tr><td>SD/30</td></tr>
<tr><td>WSD/60</td></tr>
<tr><td>HD720p/30</td><td>4</td><td>4</td></tr>
<tr><td>CIF/60</td><td rowspan="14"></td><td rowspan="6">2.66</td></tr>
<tr><td>SIF/60</td></tr>
<tr><td>WSIF/60</td></tr>
<tr><td>WCIF/60</td></tr>
<tr><td>432X336/60</td></tr>
<tr><td>480X352/60</td></tr>
<tr><td>WSD/50</td><td rowspan="5">4</td></tr>
<tr><td>4CIF/50</td></tr>
<tr><td>4SIF/60</td></tr>
<tr><td>WVGA/60</td></tr>
<tr><td>WVGA/50</td></tr>
<tr><td>HD720p/60</td><td rowspan="2">8</td></tr>
<tr><td>HD1080p/30</td></tr>
</table>

High Definition Video Switching

During a *High Definition Video Switching* conference, each endpoint uses one video (CIF) port.

Voice

One *Audio Only* resource is used to connect a single voice participants once CIF resources have been converted to *Audio Only*. However, if no CIF resources were converted, Audio Only endpoints use one CIF video resource per connection.

When video ports are fully used, the system cannot use free audio ports for video. When audio port resources are fully used, video ports can be used, using one video port to connect one voice participant.

Video/Voice Port Configuration

The *Video/Voice Port Configuration* enables you to configure the RMX Resource Capacity and if in *MPM+ System Card Configuration Mode,* to select the capacity method.

Flexible Resource Capacity Mode

All resources are initially allocated as CIF video ports as it is a resolution commonly supported by all endpoints.

The administrator can allocate some or all of these resources as *Voice* resources and let the system allocate the remaining *Video* resources automatically as participants connect to conferences. The system automatically allocates resources according to the connecting participant's endpoint type, capabilities and line rate.

If the system runs out of voice ports, voice endpoints cannot connect to available video ports. Conversely, video endpoints cannot connect to available voice ports.

Flexible Resource Capacity mode is available and is the default selection in both *MPM and MPM+ System Card Configuration Modes.* It is the only allocation method in *MPM System Card Configuration Mode*.

Fixed Resource Capacity

Fixed Resource Capacity enables the administrator to allocate the number of resources available to each video connection type and *Audio Only* connections in advance. In *Fixed Resource Capacity* mode, the system is always in a known state, and when used in conjunction with the *Resource Report,* it gives the administrator precise control over resource allocation and optimization. For more information, see "*Forcing Video Resource Allocation to CIF Resolution*" on page **16-46**.

Fixed Resource Capacity mode is available only in *MPM+ System Card Configuration Modes.*

If all resources allocated to a specific endpoint type are in use and an endpoint of that specific type tries to connect to the RMX, the RMX first attempts to connect the endpoint at the next highest resolution. If not resources are available at that level the RMX begins search for connection resolutions at progressively decreasing resolutions.

Example: In a system that has 10 SD ports allocated and in use:

If another SD endpoint (11th) attempts to connect, the system first tries to allocate resources to the SD endpoint first from HD720 and then from HD1080 resources.

If HD resources are allocated to an SD endpoint, HD endpoints may experience a resource deficiency when trying to connect and may not be connected at HD resolution.

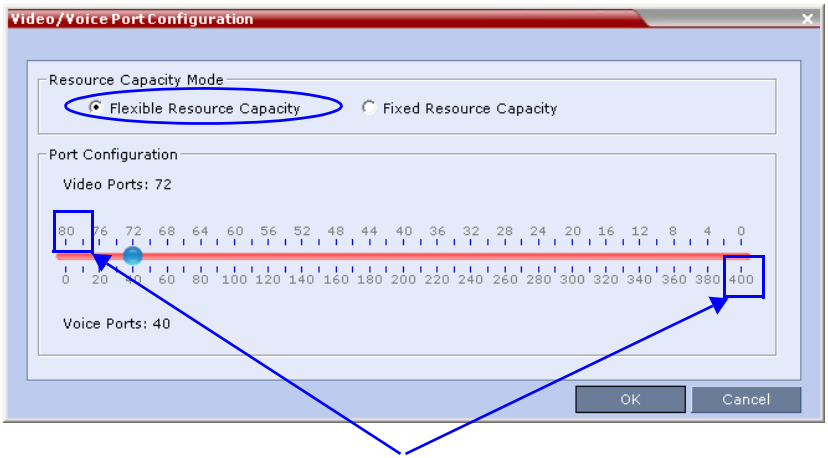
If there are no available HD resources the system tries to allocate resources to the SD endpoint from any available CIF resources.

If there are no available CIF resources the system tries to allocate resources to the SD endpoint from any available *Audio Only* resources. If *Audio Only* resources are allocated the HD endpoint, it is connected as an *Audio Only* participant.

Configuring the Video/Voice Resources in MPM Mode

To allocate Audio Only resources:

1 In the RMX menu, click **Setup > Video/Voice Port Configuration**. The *Video/Voice Port Configuration* dialog box opens.


Video/Voice Port Configuration
Resource Capacity Mode
Flexible Resource Capacity
Fixed Resource Capacity
Port Configuration
Video Ports: 72
80 76 72 68 64 60 56 52 48 44 40 36 32 28 24 20 16 12 8 4 0
0 20 40 60 80 100 120 140 160 180 200 220 240 260 280 300 320 340 360 380 400
Voice Ports: 40
OK
Cancel

Resource Maximum from License

A single slider is displayed, calibrated according to licensed video resources indicated in CIF ports in the RMX.

2 Move the slider to the number of *Audio Only* ports to be allocated.

The slider moves in multiples of two, converting CIF video ports to voice ports in groups of two, with each CIF video port converting to five voice ports. The minimum number of voice ports that can be allocated is 10 (2 video ports x 5 voice ports per video port).

3 Click **OK**.

4 Reset the MCU.

In *Flexible Resource Capacity*, any change in resource allocation requires a reset of the RMX for changes to take effect.

Configuring the Video/Voice Resources in MPM+ Mode

There are two *Resource Capacity* modes in *MPM+ Mode*:

- Flexible Resource Capacity
- Fixed Resource Capacity

The resource allocation mode is saved by the RMX and is activated when the RMX is restarted.

Flexible Resource Capacity

Flexible Resource Capacity is the default resource allocation mode in *MPM+ Mode* and is functionally identical to the *MPM Flexible Resource Capacity* described above.

To allocate Audio Only ports in MPM+ mode:

1 **Optional** *(otherwise skip to step 2)***:** If the RMX is in *Fixed Resource Capacity* mode:

   a In the RMX menu, click **Setup > Video/Voice Port Configuration**.
   The *Video/Voice Port Configuration* dialog box opens.

   b In the *Resource Capacity Mode* box, select **Flexible Resource Capacity**.

   c Click **OK**.

   d Restart the RMX from the *Hardware Monitor* pane.

2 In the RMX menu, click **Setup > Video/Voice Port Configuration**.
The *Video/Voice Port Configuration* dialog box opens.

   If switching from *Fixed* mode, all video resources are allocated as CIF video ports.

3 Continue with **Step 2** of the *MPM Mode Flexible Resource Capacity* procedure described above.

To allocate resources in Fixed Resource Capacity mode:

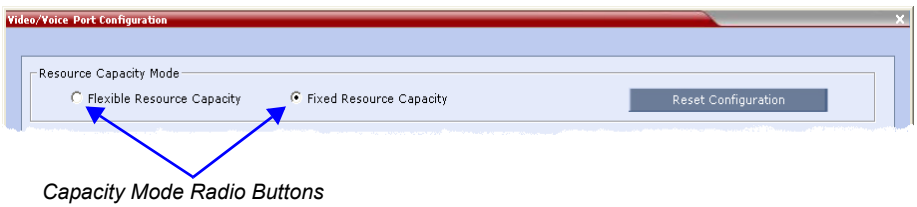
Resource re-configuration (if the system is already set to Fixed Resource Capacity mode) should only be performed when no conferences are running on the RMX.

1 **Optional** *(otherwise skip to step 2)***:** If the RMX is not in *Fixed Capacity Mode.*

a In the RMX menu, click **Setup > Video/Voice Port Configuration**.

The *Video/Voice Port Configuration* dialog box opens.

b In the *Resource Capacity Mode* box, click **Fixed**.

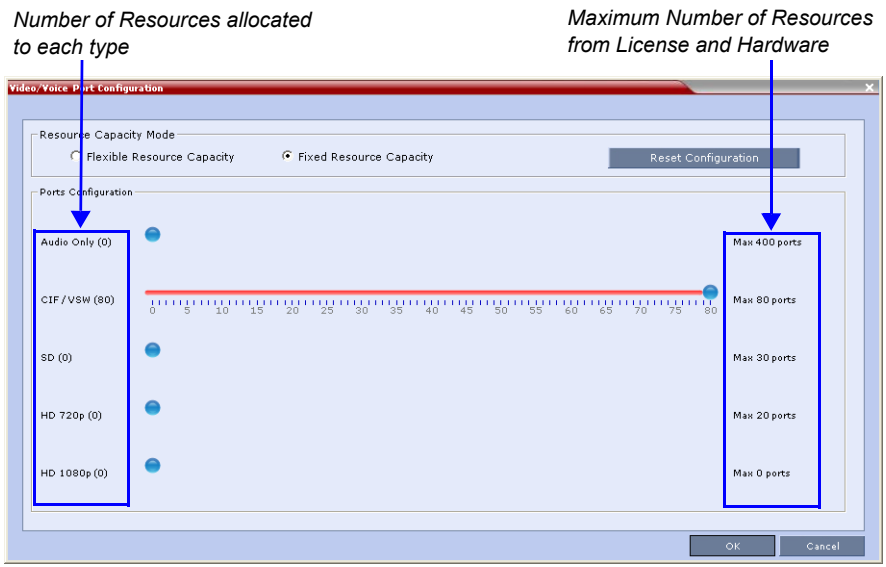


c Click **OK**.

d Restart the RMX.

2 In the RMX menu, click **Setup > Video/Voice Port Configuration**.

The *Video/Voice Port Configuration* dialog box opens.



Fixed Resource Capacity mode displays five sliders, one for each resource type: *Audio Only*, *CIF*, *SD*, *HD 720p 30fps*, *HD 1080p / HD 720p 60fps* (*HD 1080p / HD 720p 60fps* resources are represented on the same slider) where each type requires different number of video resources (in CIF ports) for connecting endpoints.

— The first time the *Fixed Resource Capacity* is selected, all resources are allocated to HD720p30 by default.
— If the allocation mode was previously *Fixed* or if it was *Auto* but *Fixed* had been selected in the past, the previous resource allocations in the mode are displayed.

The maximum number of allocatable of resources of each type for an RMX containing two fully licensed *MPM+* cards are as follows:

Resource Type	Maximum
Audio Only	800
CIF/VSW	160
SD	60
HD720p	40
HD1080p	20

The *MAX_CP_RESOLUTION* flag setting does not affect resource allocation.

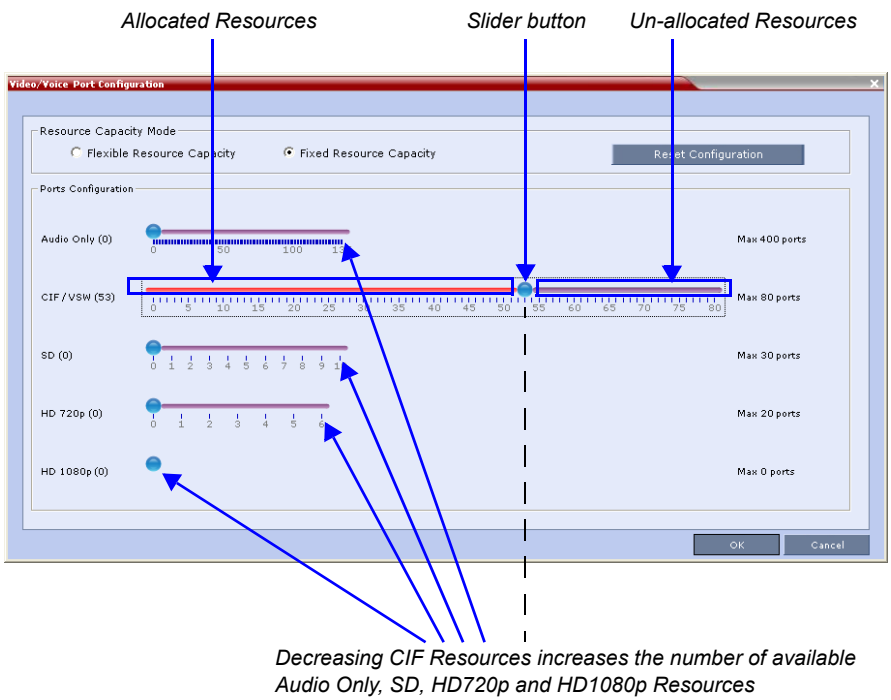
Example: If it is set to *SD30*, the *HD 1080p* slider is still displayed and *HD 1080* resources can be allocated. However, *HD 1080* participants will connect at *SD30* resolution.

Using the sliders, the administrator can manually allocate resources to the various types of video resolutions and *Audio Only* connections that can be used by connecting endpoints.

3 Move the blue slider buttons to allocate resources.

As all the resources are allocated when the dialog box opens, you must first free resources of one type by moving the blue slider button to the left, and then move blue slider button of the required resource type to the number of resources to be allocated.

On the slider bars, red areas to the left of the blue slider buttons indicate allocated resources and purple areas to the right of the blue slider buttons indicate unallocated resources in the system.



When the position of a slider is changed the system calculates the effect on the remaining system resources and adjusts the slider scales accordingly.

For example: Decreasing the allocated CIF ports from 80 to 53, free ports for allocation that can be used to allocate up to 135 voice ports or 10 SD ports or 6 HD 720p ports, or any combination of the resource types.

Allocating five *Audio Only* ports decreases the number of *CIF* ports while allocating one *SD* port decreases the number of *CIF* ports.

4 Click **OK** to activate the new *Resource Capacity*.

In *Fixed Capacity Mode* system restart is not required for the re-configuration of the allocation to take effect.

If after resources are recalculated there are purple areas to the right of the blue slider buttons indicating unallocated resources in the system, the system issues a warning stating that there are un-allocated resources in the system.

5 **Optional.** Repeat this procedure from **Step 2** to further optimize the resource allocation.

Un-allocated resources cannot be used by any participants.

If after recalculating the resources the system determines that there are insufficient resources to support the configuration indicated by the sliders:

— A major *System Alert* is raised with *Insufficient resources* in its *Description* field.

— The *Fixed Resource Capacity* blue slider buttons are disabled.

— A warning message is displayed.

- Click **OK** to close the warning message box.

a Optional.

- Click the **Reset Configuration** button to set all the blue slider buttons to zero.
- Reconfigure the resource allocation.
- Click **OK** to activate the new resource allocation.

b **Optional.** Click the **Cancel** button to accept the resource allocation.

The *System Alert* remains active.

Forcing Video Resource Allocation to CIF Resolution

You can set the MCU to allocate one CIF video resource to an endpoint, regardless of the resolution determined by the Conference Profile parameters. This forcing saves resources and enables more endpoints to connect to conferences.

The forcing is done by modifying the system configuration and it applies to all conferences running on the MCU.

You can specify the endpoint types for which resource allocation can be forced to CIF resource, enabling other types of endpoints to use higher resolutions in the same conference. For example, you can force the system to allocate one CIF video resource to CMAD and VSX endpoints while HDX endpoints can connect using SD or HD video resources.

Once the endpoint connects to the conference, its type is identified by the RMX and, if applicable, the RMX will connect it using one CIF resource, even if a higher resolution can be used.

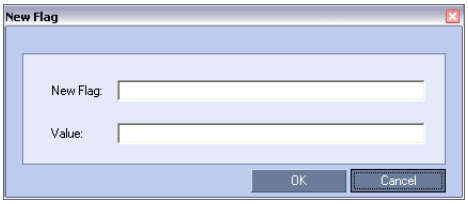
To force CIF resource:

1 On the *RMX* menu, click **Setup > System Configuration**.

The *System Flags* dialog box opens.

2 In the *MCMS_PARAMETERS* tab, click the **New Flag** button.

The *New Flag* dialog box is displayed.



3 In the *New Flag* field enter the flag name:
FORCE_CIF_PORT_ALLOCATION

4 In the *Value* field enter the product type to which the CIF resource should be allocated. Possible values are:

— **CMA Desktop** for CMA desktop client

— **VSX nnnn** where nnnn represents the model number for example, VSX 8000.

You can define several endpoint types, listing them one after the other separated by semicolon (;).
For example, CMA Desktop;VSX 8000.

5 Click **OK**.
The new flag is added to the flags list.

Reset the MCU for changes to take effect. For more details, see RMX 2000 Administrator's Guide, "*Resetting the RMX*" on page **16-90**.

To cancel the forcing of CIF resource:

1 On the *RMX* menu, click **Setup > System Configuration**.

The *System Flags* dialog box opens.

2 In the *MCMS_PARAMETERS* tab, double-click or select the flag **FORCE_CIF_PORT_ALLOCATION** and click the **Edit Flag** button.

3 In the *New Value* field, clear the value entries.

4 Click **OK**.

Reset the MCU for changes to take effect. For more details, see *RMX 2000 Administrator's Guide*, "*Resetting the RMX*" on page **16-90**.

Resource Report

The *Resource Report* displays the real time resource usage according to the selected *Resource Capacity Mode*:

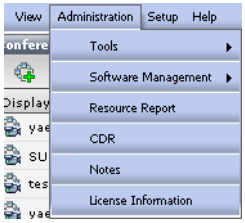
- *Flexible Resource Capacity Mode* available in both *MPM* and *MPM+ Modes*
- *Fixed Resource Capacity Mode* available only in *MPM+ Mode*

The *Resource Report* also includes a graphic representation of the resource usage.

When the RMX is working in *MPM+ Mode*, with *Fixed Resource Capacity Mode*™ selected, additional system resources information is displayed.

Displaying the Resource Report

- In the main toolbar, click **Administration > Resource Report**.



The *Resource Report* dialog box appears, displaying the resource usage according to the Resource Capacity Mode. For each resource type, the Resource Report includes the following columns:

Table 16-10 *Resource Report Fields Parameters*

Column	Description
Type	The type of audio/video resources available.

Table 16-10 Resource Report Fields Parameters (Continued)

Column	Description
Total	The *Total* column displays the total number of resources of that type as configured in the system (*Occupied* and *Free*). This number reflects the current audio/video port configuration. Any changes to the resource allocation will affect the resource usage displayed in the Resource Report.
Occupied	The number of RMX resources that are used by connected participants or reserved for defined participants.
Free	The number of RMX resources available for connecting endpoints.

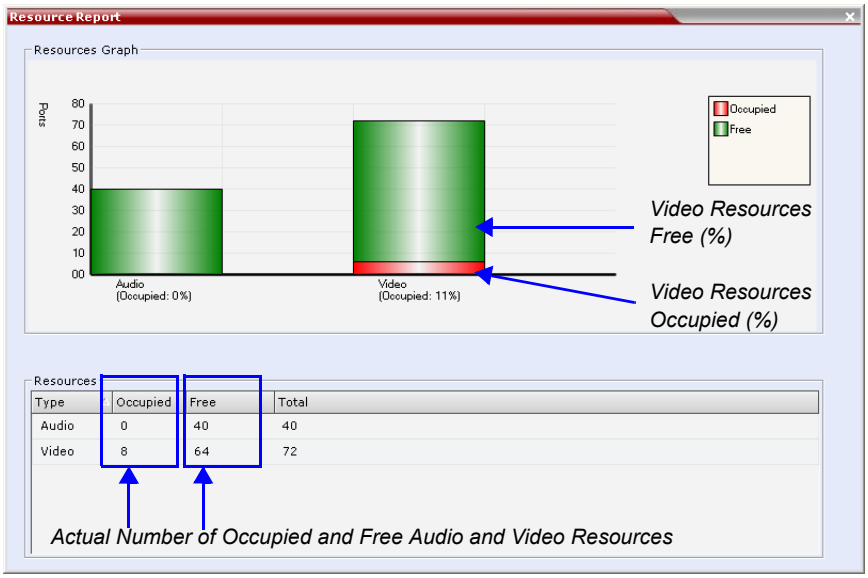
Resource Report Display in Flexible Resource Capacity Mode™

The *Resource Report* details the current availability and usage of the system resources displaying the number of free and occupied audio and video ports. A *Resources Graph* is displayed in addition to the *Resources* table.

Example: An *RMX 2000* in *Flexible Resource Capacity Mode* has:

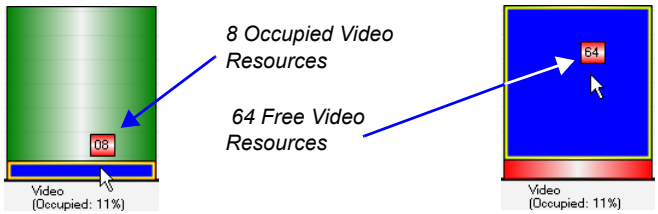
- 80 licensed *CIF* resources.
- 8 of its 80 *CIF* resources allocated as *Audio* = 40 *Audio* resources (8x5).
- All 40 *Audio* resources free (green).
- The remaining 72 *CIF* resources allocated as *Video* resources.
- 8 of the 72 *CIF* resources are occupied (red) while the remaining 64 are free.

The *Resource Report* is displayed as follows:



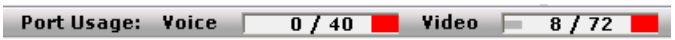
In *Flexible Resource Capacity Mode,* resource usage is displayed for *Audio* and *CIF* video resources only. They are displayed as percentages of the total resource type.

The actual number of occupied or free resources can also be displayed by moving the cursor over the columns of the bar graph. Moving the cursor over the *Video* bar displays the following:



Port Gauges

In *Flexible Resource Capacity* mode, the *Port Gauges* in the *Status Bar* show 0 of the 40 *Audio (Voice)* resources as occupied and 8 of the 72 *CIF* (*Video*) resources as occupied.



Resource Report in Fixed Resource Capacity Mode™

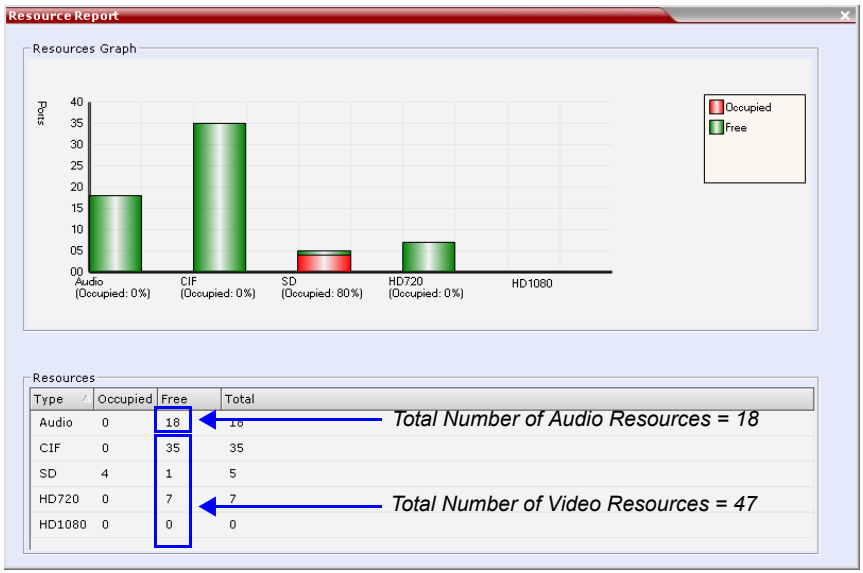
In *Fixed Resource Capacity Mode*, each resource type (*Audio, CIF, SD* and *HD*) is displayed as a bar of the graph, indicating the percentage of occupied and free resources for each resource type.

The data is also displayed as a *Resources* table indicating the actual number of resources occupied and free for each resource type along with a total number of each resource type.

Example: An *RMX 2000* in *Fixed Resource Capacity Mode* has:

- 80 licensed *CIF* resources.
- 18 *Audio* resources allocated, all free (green).
- 35 *CIF* resources allocated, all free.
- 5 *SD* resources allocated, 4 occupied (red), 1 free.
- 7 *HD 720* resources allocated, all free.
- 0 *HD 1080* resources allocated.

The *Resource Report* is displayed as follows:



The actual number of occupied or free resources can also be displayed by moving the cursor over the columns of the bar graph (as explained above for *Flexible Resource Capacity)*.

Port Gauges

Audio (*Voice*) resources are as displayed as in previous versions while all *Video* resource types are shown as a single group of *Video* resources.

The gauges show 0 of the 18 *Audio (Voice)* resources as occupied. The 4 occupied *SD* resources are shown as 4 occupied resources out of the total of 47 *Video* resources.

ISDN/PSTN

Table 16-11 lists the ISDN supported bit rates and their respective participant connection capacities per RTM ISDN card:

Table 16-11 *ISDN – E1/T1 Connection Capacity vs. Bit rate*

Bit Rates (Kbps) (Bonded)	Number of Participants per RTM ISDN Card		
	E1	T1	
128	40	40	If the conference bit rate is 128Kbps, participants connecting at bit rates lower than 128Kbps are disconnected. If the conference bit rate is above 128Kbps but does not match any of the bonded bit rates, participants are connected at the highest bonded bit rate that is less than the conference bit rate. For example: If the conference bit rate is 1024Kbps, the participant is connected at 768Kbps.
192	40	40	
256	40	40	
320	40	40	
384	34	34	
512	25	25	
768	17	17	
1152	11	11	
1472	9	9	
1536	8	8	
1920	7	6	

Port Usage

The RMX can be set to alert the administrator to potential port capacity shortages. A capacity usage threshold can be set as a percentage of the total number of licensed ports in the system.

When the threshold is exceeded, a *System Alert* is generated.

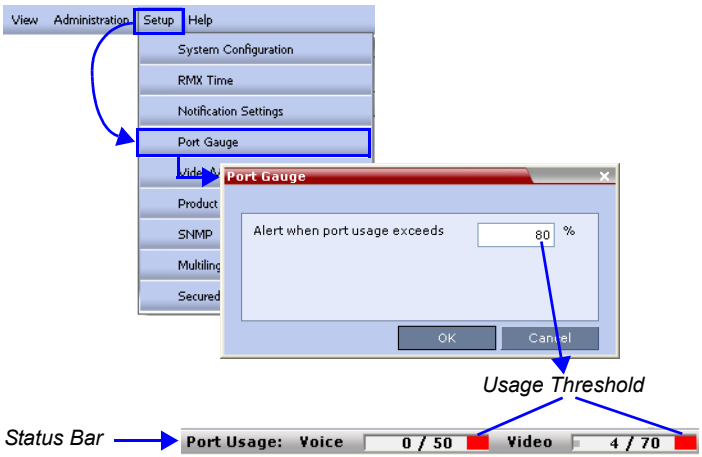
The default port capacity usage threshold is 80%.

The administrator can monitor the MCU's port capacity usage via the *Port Gauges* in the *Status Bar* of the *RMX Web Client*.

Setting the Port Usage Threshold

To Set the Port Usage Threshold:

1 In the *Setup* menu, click **Port Gauge** to open the *Port Gauge* dialog box.



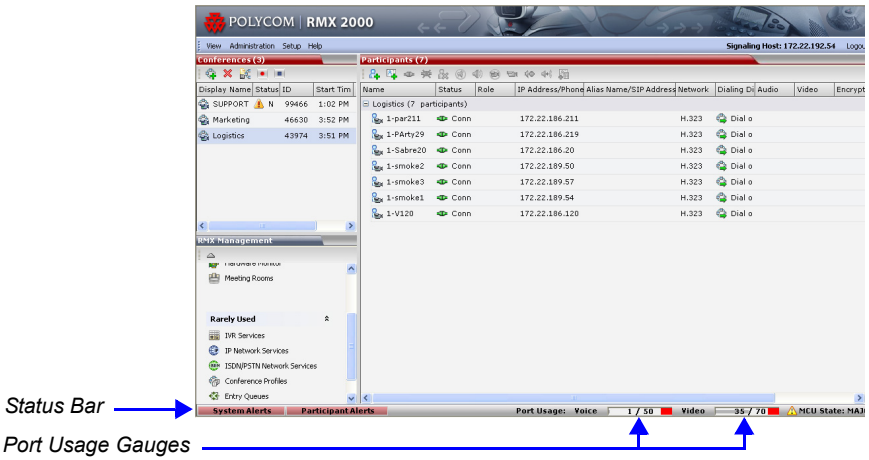
2 Enter the value for the percentage capacity usage threshold**.**

The high Port Usage threshold represents a percentage of the total number of video or voice ports available. It is set to indicate when resource usage is approaching its maximum, resulting in no free resources to run additional conferences. When port usage reaches or exceeds the threshold, the red area of the gauge flashes and a *System Alert* is generated. The default port usage threshold is 80%.

3 Click **OK.**

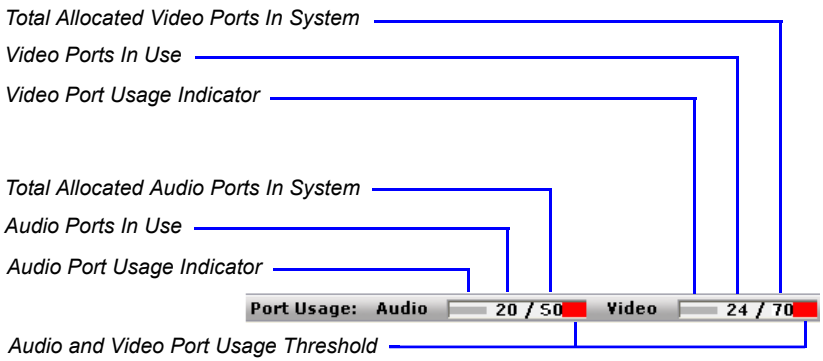
Port Usage Gauges

The *Port Usage Gauges* are displayed in the *Status Bar* at the bottom of the *RMX Web Client* screen.



The *Port Usage* gauges indicate:

- The total number of *Video* or *Voice* ports in the system according to the *Video/Voice Port Configuration*. The *Audio* gauge is displayed only if *Audio* ports were allocated by the administrator, otherwise only the *Video* port gauge is displayed.
- The number of *Video* and *Voice* ports in use.
- The *High Port Usage* threshold.


Total Allocated Video Ports In System
Video Ports In Use
Video Port Usage Indicator
Total Allocated Audio Ports In System
Audio Ports In Use
Audio Port Usage Indicator
Port Usage: Audio 20 / 50 Video 24 / 70
Audio and Video Port Usage Threshold

Port Gauges in Flexible/Fixed Capacity Modes

Audio Ports Gauge

- In both *Flexible* and *Fixed Capacity Modes:*
  The fraction displayed indicates the exact number of voice ports in use out of the total number of voice ports.

Video Ports Gauge

- In *Flexible Capacity Mode:*
  All video port usage is converted to the equivalent CIF port usage. The fraction displayed indicates the exact number of CIF video ports in use out of the total number of CIF video ports in the system.
- In *Fixed Capacity Mode:*
  All video ports are treated as a single group of *Video* resources regardless of their differing consumption of CIF ports. The fraction displayed indicates the number of video resources in use out of the total number video resources in the system.

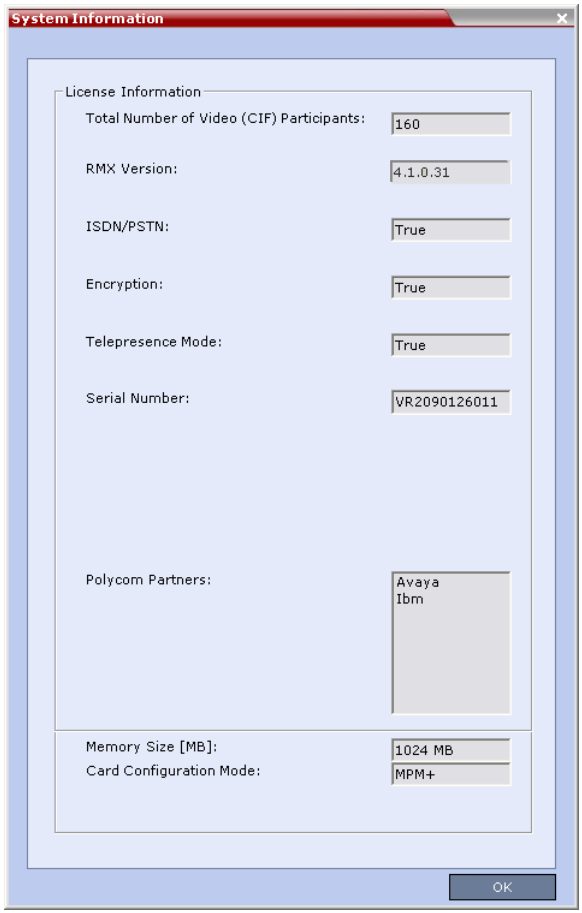
System Information

System Information includes *License Information*, and general system information, such as system memory size and *Media Card Configuration Mode*.

To view the System Information properties box:

- On the RMX menu, click **Administration > System Information**.

  The *System Information* properties box is displayed.



The *System Information* properties box displays the following information:

Table 16-12 *System Information*

Field	Description
Total Number of Video (CIF) Participants	Displays the number of CIF video participants licensed for the system.
RMX Version	Displays the *System Software Version* of the RMX.
ISDN/PSTN	The field value indicates whether RTM ISDN/ PSTN hardware has been detected in the system. Range: True / False
Encryption	The field value indicates whether *Encryption* is included in the MCU license. Encryption is not available in all countries. Range: True / False
Telepresence Mode	The field value indicates whether the system is licensed to work with *RPX* and *TPX Telepresence* room systems. Range: True / False
Serial Number	Displays the *Serial Number* of the RMX.
Polycom Partners	The field value indicates that the *System Software* contains features for the support of specific *Polycom Partner* environments.
Memory Size [MB]	This field indicates the RMX system memory size in MBytes. Possible values: • 1000 MB – The RMX can support a maximum of 800 simultaneous participant calls (if configured with two MPM+ cards). • 500 MB – The RMX can support a maximum of 400 simultaneous voice calls and 120 CIF video calls. This limitation applies to RMX's configured with either MPM or MPM+ cards.

Table 16-12 System Information (Continued)

Field	Description
Card Configuration Mode	Indicates the MCU configuration as derived from the installed media cards: • **MPM:** Only MPM cards are supported. MPM+ cards in the system are disabled. It is the mode used in previous RMX versions. • **MPM+:** Only MPM+ cards are supported. MPM cards in the system are disabled. **Note:** When started with Version 4.0 installed, the RMX enters MPM+ mode by default, even if no media cards are installed: • The RMX only switches between MPM and MPM+ *Card Configuration Modes* if MPM/MPM+ cards are removed or swapped while it is powered on. • The *Card Configuration Mode* switch occurs during the next restart. • Installing or swapping MPM/MPM+ cards while the system is off will not cause a mode switch when the system is restarted - it will restart in the *Card Configuration Mode* that was active previous to powering down.

- The RMX only switches between *MPM* and *MPM+ Card Configuration Modes* if *MPM/MPM+* cards are removed or swapped while it is running.
- The *Card Configuration Mode* switch occurs during the **next** restart.
- Installing or swapping *MPM*/*MPM+* cards while the system is off will not cause a mode switch when the system is restarted – it will restart in the *Card Configuration Mode* that was active previous to powering down.

SNMP (Simple Network Management Protocol)

SNMP standard protocol is now supported with the RMX. It enables managing and monitoring of the MCU status by **external** managing systems, such as HP OpenView or through web applications.

Detailed Description

MIBs are a collection of definitions, which define the properties of the managed object within the device to be managed. Every managed device keeps a database of values for each of the definitions written in the MIB.

The SNMP systems poll the MCU according to the MIB definitions. In addition, the MCU is able to send Traps to different managers. Traps are messages that are sent by the MCU to the SNMP Manager when an event such as MCU Reset occurs.

MIB (Management Information Base) Files

The H.341 standard defines the MIBs that H.320 and H.323 MCUs must comply with. In addition, other MIBs should also be supported, such as MIB-II and the ENTITY MIB, which are common to all network entities.

The MIBS are contained in files in the *SNMP MIBS* sub-directory of the RMX root directory. The files should be loaded to the SNMP external system and compiled within that application. Only then can the SNMP external application perform the required monitoring tasks.

The MULTI-MEDIA_MIB_TC must be compiled before compiling the other MIBs.

Private MIBS

- *RMX-MIB (RMX-MIB.MIB)*
  - — Contains the statuses of the RMX: Startup, Normal and Major.
  - — Contains all the Alarms of the RMX that are sent to the SNMP Manager.

Support for MIB-II Sections

The following table details the MIB-II sections that are supported:

Table 16-13 Supported MIB-II Sections

Section	Object Identifier
system	mib-2 1
interfaces	mib-2 2
ip	mib-2 4

The Alarm-MIB

MIB used to send alarms. When a trap is sent, the Alarm-MIB is used to send it.

H.341-MIB (H.341 – H.323)

- Gives the address of the gatekeeper.
- Supports H.341-MIB of SNMP events of H.323.

Standard MIBs

This section describes the MIBs that are included with the RMX. These MIBs define the various parameters that can be monitored, and their acceptable values.

MIB Name	Description
MULTI-MEDIA-MIB-TC (MULTIMTC.MIB)	Defines a set of textual conventions used within the set of Multi Media MIB modules.
H.320ENTITY-MIB (H320-ENT.MIB)	This is a collection of common objects, which can be used in an H.320 terminal, an H.320 MCU and an H.320/H.323 gateway. These objects are arranged in three groups: Capability, Call Status, and H.221 Statistics.

MIB Name	Description
H.320MCU-MIB (H320-MCU.MIB)	Used to identify managed objects for an H.320 MCU. It consists of four groups: System, Conference, Terminal, and Controls. The *Conference* group consists of the active conferences. The *Terminal* group is used to describe terminals in active MCU conferences. The *Controls* group enables remote management of the MCU.
H323MC-MIB (H323-MC.MIB)	Used to identify objects defined for an H.323 Multipoint Controller. It consists of six groups: System, Configuration, Conference, Statistics, Controls and Notifications. The *Conference* group is used to identify the active conferences in the MCU. The *Notifications* group allows an MCU, if enabled, to inform a remote management client of its operational status.
MP-MIB (H323-MP.MIB)	Used to identify objects defined for an H.323 Multipoint Processor, and consists of two groups: Configuration and Conference. The *Configuration* group is used to identify audio/video mix configuration counts. The *Conference* group describes the audio and video multi-processing operation.
MIB-II/RFC1213-MIB (RFC1213.MIB)	Holds basic network information and statistics about the following protocols: TCP, UDP, IP, ICMP and SNMP. In addition, it holds a table of interfaces that the Agent has. MIB-II also contains basic identification information for the system, such as, Product Name, Description, Location and Contact Person.
ENTITY-MIB (ENTITY.MIB)	Describes the unit physically: Number of slots, type of board in each slot, and number of ports in each slot.

Traps

Three types of traps are sent as follows:

1 ColdStart trap. This is a standard trap which is sent when the MCU is reset.

```
coldStart notification received from: 172.22.189.154 at 5/20/
  2007 7:03:12 PM
  Time stamp: 0 days 00h:00m:00s.00th
  Agent address: 172.22.189.154 Port: 32774 Transport: IP/UDP
  Protocol: SNMPv2c Notification
  Manager address: 172.22.172.34 Port: 162 Transport: IP/UDP
  Community: public
  Enterprise: enterprises.8072.3.2.10
  Bindings (3)
    Binding #1: sysUpTime.0 *** (timeticks) 0 days
    00h:00m:00s.00th
    Binding #2: snmpTrapOID.0 *** (oid) coldStart
    Binding #3: snmpTrapEnterprise.0 *** (oid)
    enterprises.8072.3.2.10
```

Figure 1 An Example of a ColdStart Trap

2 Authentication failure trap. This is a standard trap which is sent when an unauthorized community tries to enter.

```
authentication Failure notification received from:
  172.22.189.154 at 5/20/2007 7:33:38 PM
  Time stamp: 0 days 00h:30m:27s.64th
  Agent address: 172.22.189.154 Port: 32777 Transport: IP/UDP
  Protocol: SNMPv2c Notification
  Manager address: 172.22.172.34 Port: 162 Transport: IP/UDP
  Community: public
  Enterprise: enterprises.8072.3.2.10
  Bindings (3)
    Binding #1: sysUpTime.0 *** (timeticks) 0 days
    00h:30m:27s.64th
    Binding #2: snmpTrapOID.0 *** (oid) authenticationFailure
    Binding #3: snmpTrapEnterprise.0 *** (oid)
    enterprises.8072.3.2.10
```

Figure 2 An Example of an Authentication Failure Trap

3 Alarm Fault trap. The third trap type is a family of traps defined in the POLYCOM-RMX-MIB file, these traps are associated with the RMX active alarm and clearance (proprietary SNMP trap).

```
rmxFailedConfigUserListInLinuxAlarmFault notification received
  from: 172.22.189.154 at 5/20/2007 7:04:22 PM
  Time stamp: 0 days 00h:01m:11s.71th
  Agent address: 172.22.189.154 Port: 32777 Transport: IP/UDP
  Protocol: SNMPv2c Notification
  Manager address: 172.22.172.34 Port: 162 Transport: IP/UDP
  Community: public
  Bindings (6)
    Binding #1: sysUpTime.0 *** (timeticks) 0 days
    00h:01m:11s.71th
    Binding #2: snmpTrapOID.0 *** (oid)
    rmxFailedConfigUserListInLinuxAlarmFault
    Binding #3: rmxAlarmDescription *** (octets) Insufficient
    resources
    Binding #4: rmxActiveAlarmDateAndTime *** (octets) 2007-6-
    19,16:7:15.0,0:0
    Binding #5: rmxActiveAlarmIndex *** (gauge32) 2
    Binding #6: rmxActiveAlarmListName *** (octets) Active
    Alarm Table
 *  Binding #7: rmxActiveAlarmRmxStatus *** (rmxStatus) major
```

Figure 3 *An Example of an Alarm Fault Trap*

Each trap is sent with a time stamp, the agent address and the manager address.

Status Trap Content

The MCU sends status traps for the status **MAJOR** - a trap is sent when the card/MCU status is MAJOR.

All trap content is considered "MAJOR".

Defining the SNMP Parameters in the RMX

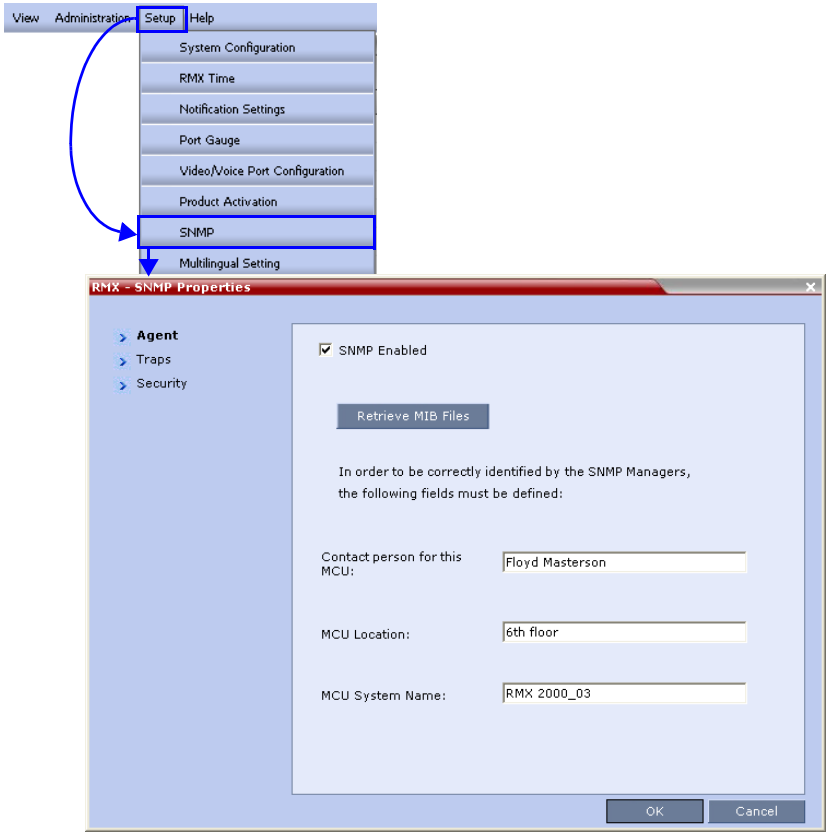
The SNMP option is enabled via the RMX Web Client application.

The addresses of the Managers monitoring the MCU and other security information are defined in the RMX Web Client application and are saved on the MCU's hard disk. Only users defined as Administrator can define or modify the SNMP security parameters in the RMX Web Client application.

To enable SNMP option:

1 In the RMX Web Client menu bar, click **Setup>SNMP**.

The *RMX-SNMP Properties - Agent* dialog box is displayed.



This dialog box is used to define the basic information for this MCU that will be used by the SNMP system to identify it.

2 In the *Agent* dialog box, click the **SNMP Enabled** check box.

3 Click the **Retrieve MIB Files** button to obtain a file that lists the MIBs that define the properties of the object being managed.

The *Retrieve MIB Files* dialog box appears.

4 Click the **Browse** button and navigate to the desired directory to save the MIB files.

5 Click **OK**.
The path of the selected directory is displayed in the *Retrieve MIB Files* dialog box.

6 Click the **Save** button.
The MIB files are saved to the selected directory.

7 Click Close to exit the *Retrieve MIB Files* dialog box.

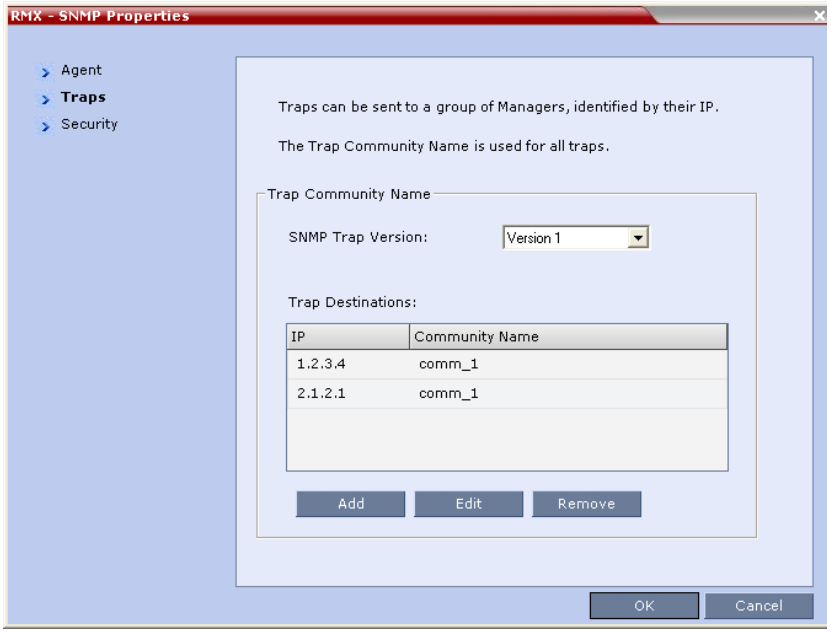
8 In the *Agent* dialog box, define the parameters that allow the SNMP Management System and its user to easily identify the MCU.

***Table 16-14** RMX-SNMP Properties - Agent Options*

Field	Description
Contact person for this MCU	Type the name of the person to be contacted in the event of problems with the MCU.
MCU Location	Type the location of the MCU (address or any description).
MCU System Name	Type the MCU's system name.

9 Click the **Traps** tab.

The *RMX-SNMP Properties – Traps* dialog box opens.



Traps are messages sent by the MCU to the SNMP Managers when events such as MCU Startup or Shutdown occur. Traps may be sent to several SNMP Managers whose IP addresses are specified in the *Trap Destinations* box.

10 Define the following parameters:

***Table 16-15** RMX-SNMP Properties – Traps Options*

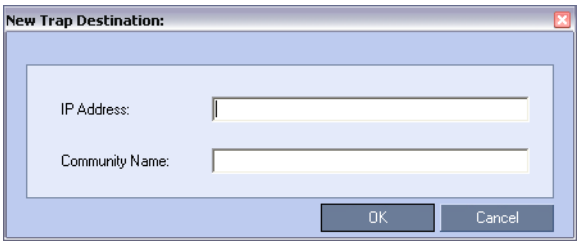
Field	Description
SNMP Trap Version	Specifies the version, either *Version1* or *Version2c*, of the traps being sent to the IP Host. Polycom software supports the standard SNMP version 1 and 2 traps, which are taken from RFC 1215, convention for defining traps for use with SNMP: **Note:** The *SNMP Trap Version* parameters must be defined identically in the external SNMP application.

Table 16-15 RMX-SNMP Properties – Traps Options (Continued)

Field	Description
Trap Destination	This box lists the currently defined IP addresses of the Manager terminals to which the message (trap) is sent.

11 Click the **Add** button to add a new Manager terminal.

The *New Trap Destination* dialog box opens.



12 Type the **IP Address** and the **Community name** of the manager terminal used to monitor the MCU activity, and then click **OK**.

The *Community name* is a string of characters that will be added to the message that is sent to the external Manager terminals. This string is used to identify the message source by the external Manager terminal.

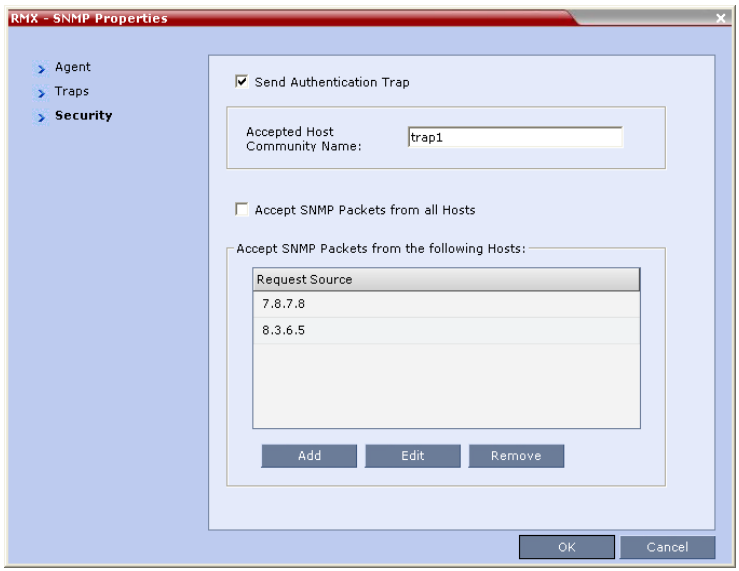
The new *IP Address* and *Community name* is added to the *Trap Destinations* box.

a To delete the IP Address of a Manager terminal, select the address that you wish to delete, and then click the **Remove** button.

The IP address in the *Trap Destinations* box is removed.

13 Click the **Security** tab.

The *RMX-SNMP Properties – Security* dialog box opens.



This dialog box is used to define whether the query sent to the MCU is sent from an authorized source. When the *"Accept SNMP packets from all Hosts"* is disabled, a valid query must contain the appropriate community string and must be sent from one of the Manager terminals whose IP address is listed in this dialog box.

14 Define the following parameters:

***Table 16-16** RMX-SNMP Properties – Security Options*

Field	Description
Send Authentication Trap	Select this check box to send a message to the SNMP Manager when an unauthorized query is sent to the MCU. When cleared, no indication will be sent to the SNMP Manager.
Accept Host Community Name	Type the string added to queries that are sent from the SNMP Manager to indicate that they were sent from an authorized source.

Table 16-16 RMX-SNMP Properties – Security Options (Continued)

Field	Description
Accept Host Community Name (cont.)	**Note:** Queries sent with different strings will be regarded as a violation of security, and, if the *Send Authentication Trap* check box is selected, an appropriate message will be sent to the SNMP Manager.
Accept SNMP Packets from all Host	Select this option if a query sent from any Manager terminal is valid. When selected, the *Accept SNMP Packets from These Hosts* option is disabled.
Accept SNMP Packets from the following Hosts	Lists specific Manager terminals whose queries will be considered as valid. This option is enabled when the *Accept SNMP Packets from any Host* option is cleared.

15 To specifically define one or more valid terminals, ensure that the *Accept SNMP Packets from any Host* option is cleared and then click the **Add** button.

The *Accepted Host IP Address* dialog box opens.

16 Enter the *IP Address* of the Manager terminal from which valid queries may be sent to the MCU, and then click **OK**.
Click the **Add** button to define additional *IP Addresses*.
The *IP Address* or *Addresses* are displayed in the *Accept SNMP Packets from These Hosts* box.

Queries sent from terminals not listed in the *Accept SNMP Packets from These Hosts* box are regarded as a violation of the MCU security, and if the *Send Authentication Trap* check box is selected, an appropriate message will be sent to all the terminals listed in the *SNMP Properties – Traps* dialog box.

17 In the *RMX - SNMP Properties - Security* dialog box, click **OK**.

Multilingual Setting

Each supported language is represented by a country flag in the *Welcome Screen* and can be selected as the language for the *RMX Web Client*.

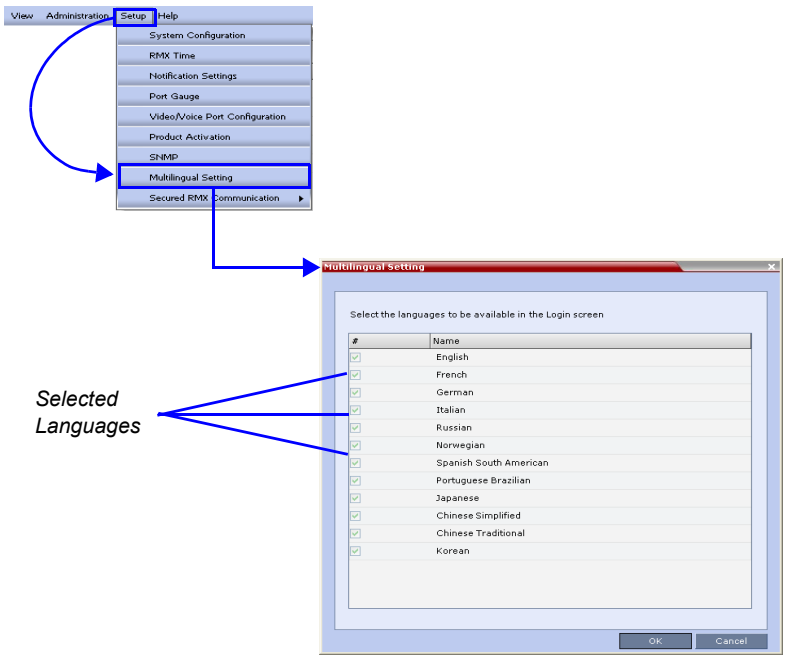
Customizing the Multilingual Setting

The number of languages available for selection in the *Login* screen of the *RMX Web Client* can be modified by selecting the *Setup* > *Multilingual Setting* option from the RMX menu.

To customize the Multilingual Setting:

1 On the RMX menu, click **Setup > Multilingual Setting**.

The *Multilingual Setting* dialog box is displayed.



2 Click the check boxes of the languages to be available for selection.

3 Click **OK**.

4 **Log out** from the RMX Web Client and **Log in** for the customization to take effect**.**

Software Management

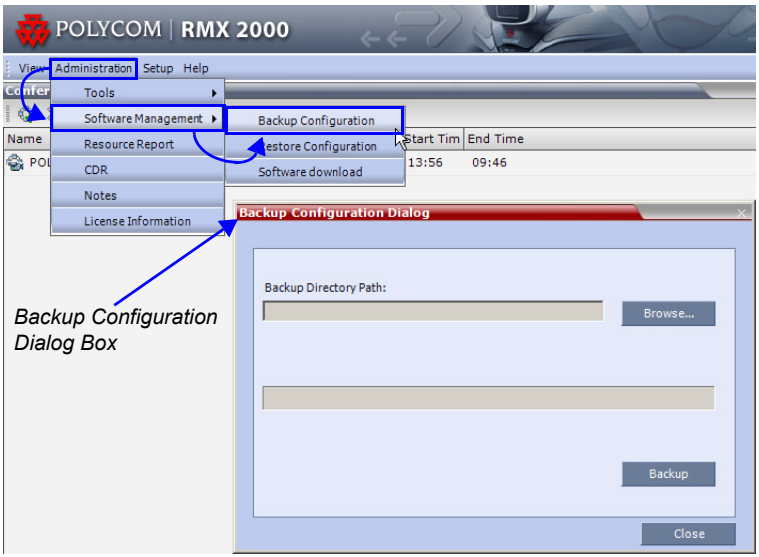
The *Software Management* menu is used to backup and restore the RMX's configuration files and to download MCU software.

Using Software Management

To backup configuration files:

1 On the *RMX* menu, click **Administration > Software Management > Backup Configuration**.

The *Backup Configuration* dialog box opens.



2 **Browse** to the *Backup Directory Path* and then click **Backup**.

To restore configuration files:

1. On the *RMX* menu, click **Administration > Software Management > Restore Configuration**.
2. **Browse** to the *Restore Directory Path* where the backed up configuration files are stored and then click **Restore**.

To download MCU software files:

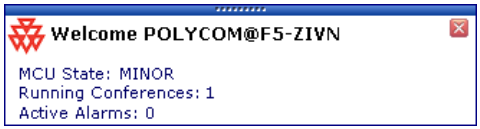
1. On the *RMX* menu, click **Administration > Software Management > Software Download**.
2. **Browse** to the *Install Path* and then click **Install**.

Notification Settings

The RMX can display notifications when:

- A new RMX user connects to the MCU.
- A new conference is started.
- Not all defined participants are connected to the conference or when a single participant is connected
- A change in the MCU status occurs and an alarm is added to the alarm's list.

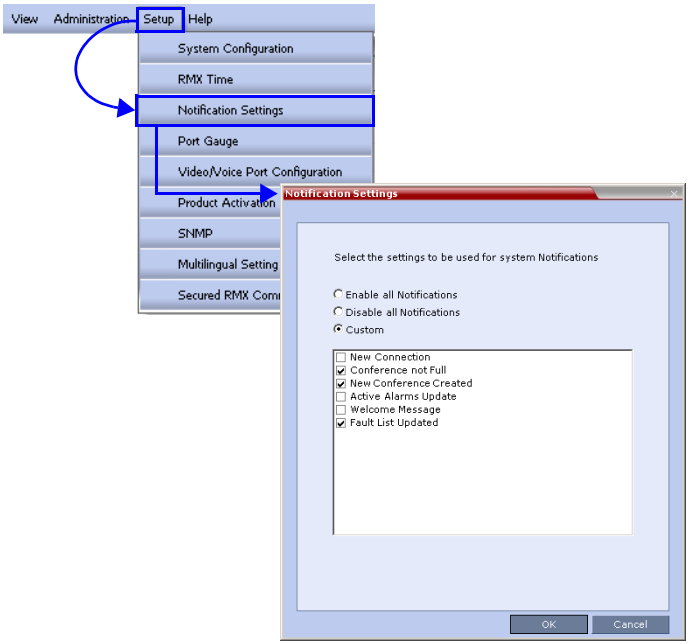
A welcome message is displayed to the RMX user upon connection.



To configure the notifications:

1 On the *RMX* menu, select **Setup > Notification Settings**.

The *Notification Settings* dialog box appears.



The following notification options are displayed.

Table 16-17 Notification Settings Parameters

Field	Description
New Connection	Notification of a new user/administrator connecting to the RMX
New Conference Created	New conference has been created.
Conference Not Full	The conference is not full and additional participants are defined for the conference.
Welcome Message	A welcome message after user/administrator logon.
Active Alarms Update	Updates you of any new alarm that occurred.
Fault List Updated	Updates you when the faults list is updated (new faults are added or existing faults are removed).

2 **Enable/Disable All Notifications** or **Custom** to select specific notifications to display.

3 Click **OK**.

Logger Diagnostic Files

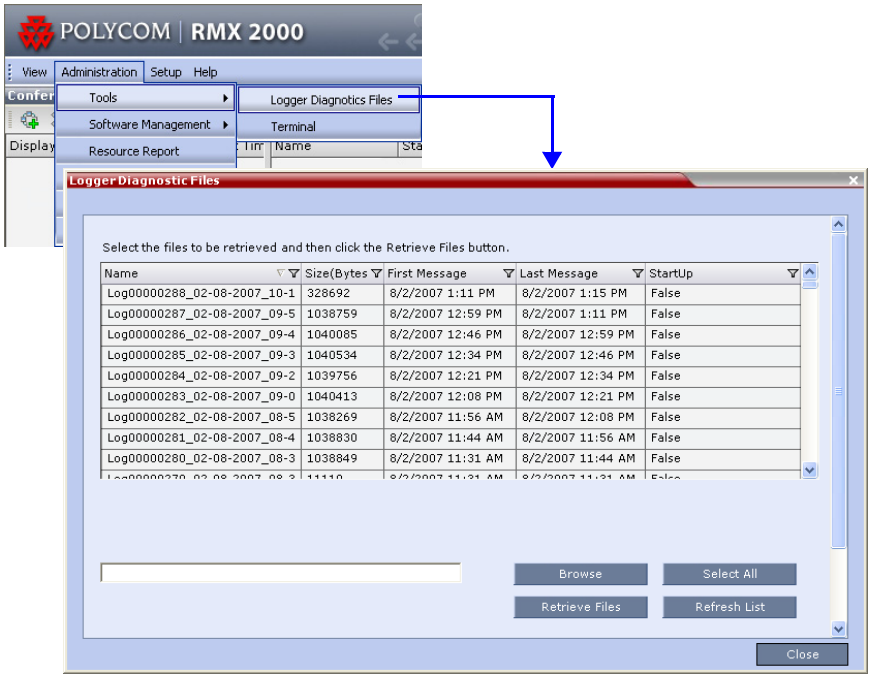
The Logger utility is a troubleshooting tool that continually records MCU system messages and saves them to files in the MCU hard drive. For each time interval defined in the system, a different data file is created. The files may be retrieved from the hard drive for off-line analysis and debugging purposes.

The Logger utility is activated at the MCU startup. The Logger is disabled when the MCU is reset manually or when there is a problem with the Logger utility, e.g. errors on the hard drive where files are saved. In such cases, data cannot be retrieved.

When the MCU is reset via the RMX, the files are saved on the MCU hard drive.

To access the Logger Diagnostic Files:

- On the *RMX* menu, click **Administration > Tools > Logger Diagnostic Files**.



The following tasks can be performed:

Table 16-18 Diagnostic File Button Options

Button	Description
Refresh List	Refreshes the list and adds newly generated logger files.
Select All	Selects all the logger files listed.
Browse	Selects the destination folder for download.
Retrieve Files	Saves files to the destination folder.

When retrieved, the log file name structure is as follows:

- Sequence number (starting with 1)
- Date and Time of first message
- Date and Time of last message
- File size
- Special information about the data, such as Startup

File name structure:

Log_SNxxxxxxxxxx_FMDddmmyyy_FMThhmm_LMDddmmyyyy_LMThhmm_SZxxxxxxxxxx_SUY.log

File name format:

- SN = Sequence Number
- FM = First Message, date and time
- LM = Last Message, date and time
- SZ = Size
- SU = Startup (Y/N) during the log file duration

Example:

Log_SN0000000002_FMD06032007_FMT083933_LMD06032007_LMT084356_SZ184951_SUY.log.

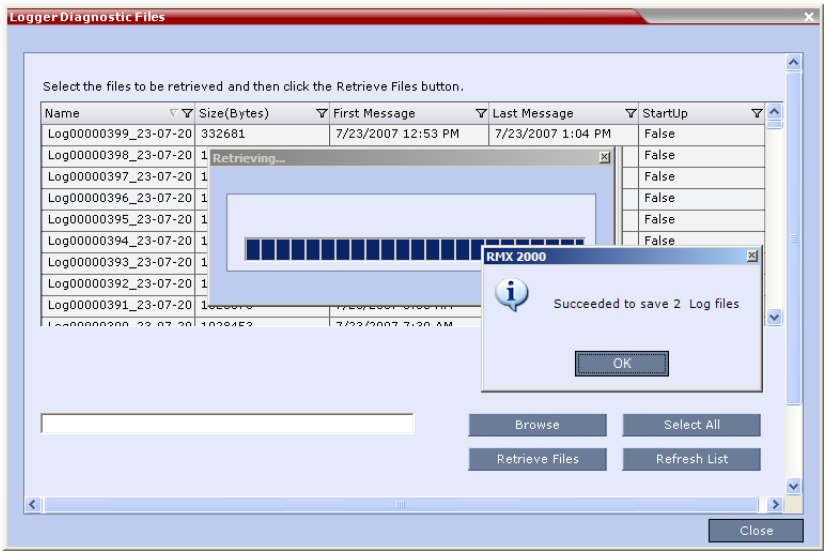
Retrieving the Logger Files:

1. Select the log files to retrieve. Multiple selections of files are enabled using standard Windows conventions.
2. In the *Logger Diagnostic Files* dialog box, click the **Browse** button.

3 In the *Browse for Folder* window, select the directory location to save the Logger files and click **OK**.

You will return to the *Logger Diagnostic Files* dialog box.

4 Click the **Retrieve Files** button.



The log files (in *.txt format) are saved to the defined directory and a confirmation caption box appears indicating a successful retrieval of the log files.

Viewing the Logger File contents:

To analyze the log files generated by the system, open the retrieved *.txt files in any text editor application, i.e. Notepad, Textpad or MS Word.

1 Using Windows Explorer, browse to the directory containing the retrieved log files.

2 Use any text editor application to open the log file(s).

Auditor

The *Event Auditor* enables administrators to analyze configuration changes and unusual or malicious activities in the RMX system.

A new user type, *Auditor,* has also been created. An *Auditor* is a user that can view *Auditor Files* and audit the system.

Auditor operates in real time, recording all administration activities and login attempts from the following RMX modules:

- Control Unit
- Shelf Manager

For a full list of monitored activities, see Table 16-20 on page **16-84** and Table 16-21 on page **16-85**.

The *Auditor* must always be active in the system. A *System Alert* is displayed if it becomes inactive for any reason.

The *Auditor* tool is composed of the *Auditor Files* and the *Auditor File Viewer* that enables you to view the *Auditor Files*.

Auditor Files

Auditor Event History File Storage

All audit events are saved to a buffer file on hard disk in real time and then written to a file on hard disk in XML in an uncompressed format.

A new current auditor event file is created when:

- the system is started
- the size of the current auditor event file exceeds 2 MB
- the current auditor event file's age exceeds 24 hours

Up to 1000 auditor event files are stored per RMX. These files are retained for at least one year and require 1.05 GB of disk space. The files are automatically deleted by the system (oldest first) when the system reaches the auditor event file limit of 1000.

A *System Alert* is displayed with *Can't store data* displayed in its *Description* field if:

- the system cannot store 1000 files
- the RMX does not have available disk space to retain files for one year

Audit Event Files are retained by the RMX for at least 1 year. Any attempt to delete an audit event file that is less than one year old raises a *System Alert* with *File was removed* listed in the *Description* field.

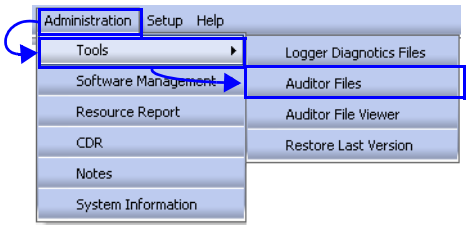
Using the *Restore Factory Defaults* of the *System Restore* procedure erases *Audit Files*.

Retrieving Auditor Files

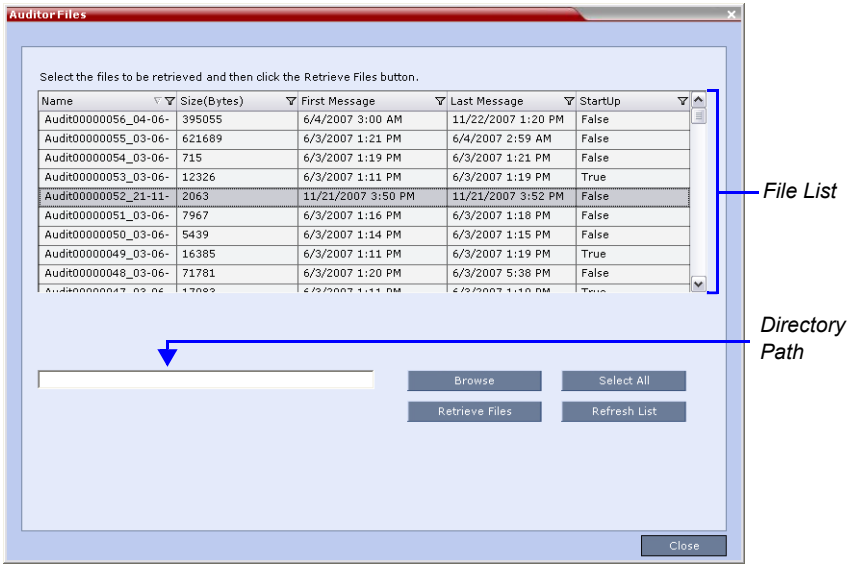
You can open the *Auditor* file directly from the *Auditor Files* list or you can retrieve the files and save them to a local workstation.

To access Auditor Files:

1 On the *RMX* menu, click **Administration > Tools > Auditor Files**.



The *Auditor Files* dialog box is displayed.



The *Auditor Files* dialogue box displays a file list containing the following file information:

- *Name*
- *Size (Bytes)*
- *First Message* – date and time of the first audit event in the file
- *Last Message* – date and time of the last audit event in the file
- *StartUp:*
  - *True* – file was created when the system was started
  - *False* – file was created when previous audit event file reached a size of 2 MB or was more than 24 hours old

The order of the *Auditor Files* dialog box field header columns can be changed and the fields can be filtered to enable searching.

For more information, see "*Auditor File Viewer*" on page **16-81**.

To retrieve files for storage on a workstation:

1. Click **Browse** and select the folder on the workstation to receive the files and then click **OK**.

   The folder name is displayed in the directory path field.

2. Select the file(s) to be retrieved by clicking their names in the file list or click **Select All** to retrieve all the files. (Windows multiple selection techniques can be used.)

3. Click **Retrieve Files.**

   The selected files are copied to the selected directory on the workstation.

To open the file in the Auditor File Viewer:

- Double-click the file.

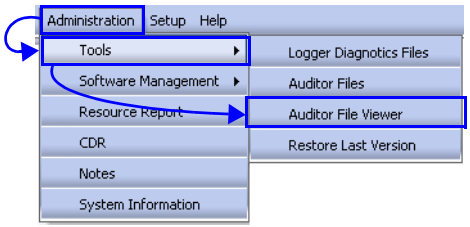
Auditor File Viewer

The *Auditor File Viewer* enables *Auditors* and *Administrators* to view the content of and perform detailed analysis on auditor event data in a selected *Auditor Event File*.

You can view an *Auditor Event File* directly from the *Auditor Files* list or by opening the file from the *Auditor File Viewer*.

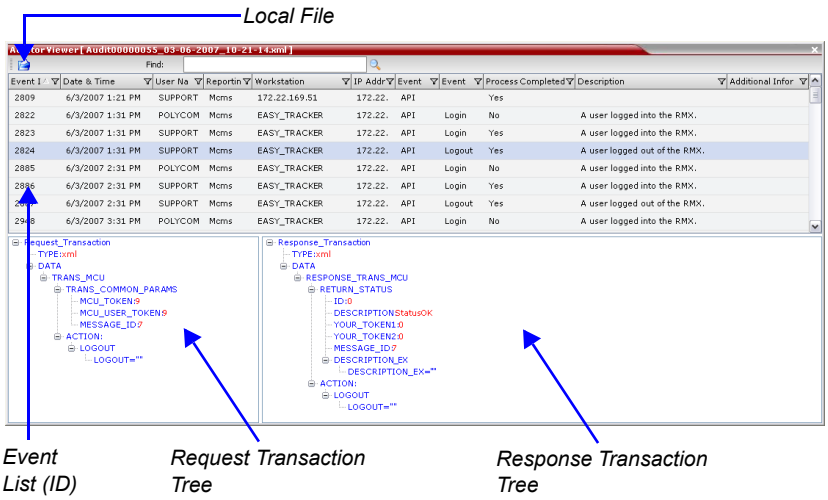
To open the Auditor Viewer from the Administration Menu:

1 On the *RMX* menu, click **Administration > Tools > Auditor File Viewer**.



The *Auditor File Viewer* is displayed.

If you previously double clicked an *Auditor Event File* in the *Auditor Files* list, that file is automatically opened.



The following fields are displayed for each event:

Table 16-19 Auditor Event Columns

Field	Description
Event ID	The sequence number of the event generated by the RMX.
Date & Time	The date and time of the event taken from the RMX's *Local Time* setting.
User Name	The *Username* (Login Name) of the user that triggered the event.
Reporting Module	The RMX system internal module that reported the event: • MCMS • MPL • Central Signaling • MPL Simulation • RMX Web Client • CM Switch • Shelf Management • ART • Video • Card Manager • RTM • MUX
Workstation	The name (alias) of the workstation used to send the request that triggered the event.
IP Address (Workstation)	The IP address of the workstation used to send the request that triggered the event.
Event Type	Auditor events can be triggered by: • API • HTTP • RMX Internal Event

Table 16-19 Auditor Event Columns (Continued)

Field	Description
Event	The process, action, request or transaction that was performed or rejected. • POST:SET transactions (API) • Configuration changes via XML (API) • Login/Logout (API) • GET (HTTP) • PUT (HTTP) • MKDIR (HTTP) • RMDIR (HTTP) • Startup (RMX Internal Event) • Shutdown (RMX Internal Event) • Reset (RMX Internal Event) • Enter Diagnostic Mode (RMX Internal Event) • IP address changes via USB (RMX Internal Event)
Process Completed	Status of the process, action, request or transaction returned by the system: • Yes – performed by the system. • No – rejected by the system.
Description	A text string describing the process, action, request or transaction.
Additional Information	An optional text string describing the process, action, request or transaction in additional detail.

The order of the *Auditor File Viewer* field header columns can be changed and the fields can be sorted and filtered to facilitate different analysis methods.

2 In the event list, click the events or use the keyboard's Up-arrow and Down-arrow keys to display the *Request Transaction* and *Response Transaction* XML trees for each audit event.

The transaction XML trees can be expanded and collapsed by clicking the expand
(⊞) and collapse (⊟) buttons.

To open an auditor event file stored on the workstation:

1. Click the **Local File** button ( ) to open the *Open* dialogue box.
2. Navigate to the folder on the workstation that contains the audit event file.
3. Select the audit event file to be opened.
4. Click **Open**.

   The selected file is opened in the *Auditor Viewer*.

Audit Events

Alerts and Faults

Table 1 lists *Alerts* and *Faults* that are recorded by the *Auditor*.

Table 16-20 *Alerts and Faults*

Event
A new activation key was loaded.
A new version was installed.
A private version is loaded.
Central Signaling indicating Recovery status.
Failed to open Apache server configuration file.
Failed to save Apache server configuration file.
Fallback version is being used.
File system scan failure.
File system space shortage.
Internal MCU reset.
Internal System configuration during startup.
Invalid date and time.
Invalid MCU Version.

Table 16-20 Alerts and Faults (Continued)

Event
IP addresses of Signaling Host and Control Unit are the same.
IP Network Service configuration modified.
IP Network Service deleted.
MCU reset.
MCU Reset to enable Diagnostics mode.
Music file error.
NTP synchronization failure.
Polycom default User exists.
Restoring Factory Defaults.
Secured SIP communication failed.
SSH is enabled.
System Configuration modified.
System is starting.
Terminal initiated MCU reset.
The Log file system is disabled.
The software contains patch(es).
User initiated MCU reset.

Transactions

Table 2 lists *Transactions* that are recorded by the *Auditor*.

Table 16-21 Transactions

Transaction
TRANS_CFG:SET_CFG
TRANS_IP_SERVICE:DEL_IP_SERVICE

Table 16-21 Transactions (Continued)

Transaction
TRANS_IP_SERVICE:NEW_IP_SERVICE
TRANS_IP_SERVICE:SET_DEFAULT_H323_SERVICE
TRANS_IP_SERVICE:SET_DEFAULT_SIP_SERVICE
TRANS_IP_SERVICE:UPDATE_IP_SERVICE
TRANS_IP_SERVICE:UPDATE_MANAGEMENT_NETWORK
TRANS_ISDN_PHONE:ADD_ISDN_PHONE
TRANS_ISDN_PHONE:DEL_ISDN_PHONE
TRANS_ISDN_PHONE:UPDATE_ISDN_PHONE
TRANS_ISDN_SERVICE:DEL_ISDN_SERVICE
TRANS_ISDN_SERVICE:NEW_ISDN_SERVICE
TRANS_ISDN_SERVICE:SET_DEFAULT_ISDN_SERVICE
TRANS_ISDN_SERVICE:UPDATE_ISDN_SERVICE
TRANS_MCU:BEGIN_RECEIVING_VERSION
TRANS_MCU:COLLECT_INFO
TRANS_MCU:CREATE_DIRECTORY
TRANS_MCU:FINISHED_TRANSFER_VERSION
TRANS_MCU:LOGIN
TRANS_MCU:LOGOUT
TRANS_MCU:REMOVE_DIRECTORY
TRANS_MCU:REMOVE_DIRECTORY_CONTENT
TRANS_MCU:RENAME
TRANS_MCU:RESET
TRANS_MCU:SET_PORT_CONFIGURATION
TRANS_MCU:SET_RESTORE_TYPE

Table 16-21 Transactions (Continued)

Transaction
TRANS_MCU:SET_TIME
TRANS_MCU:TURN_SSH
TRANS_MCU:UPDATE_KEY_CODE
TRANS_OPERATOR:CHANGE_PASSWORD
TRANS_OPERATOR:DELETE_OPERATOR
TRANS_OPERATOR:NEW_OPERATOR
TRANS_RTM_ISDN_SPAN:UPDATE_RTM_ISDN_SPAN
TRANS_SNMP:UPDATE

ActiveX Bypass

At sites that, for security reasons, do not permit Microsoft® ActiveX® to be installed, the MSI (Windows Installer File) utility can be used to install .NET Framework and .NET Security Settings components on workstations throughout the network.

All workstation that connect to RMX systems must have both .NET Framework and .NET Security Settings running locally. These components are used for communication with the RMX and can only be installed on workstations by users with administrator privileges.

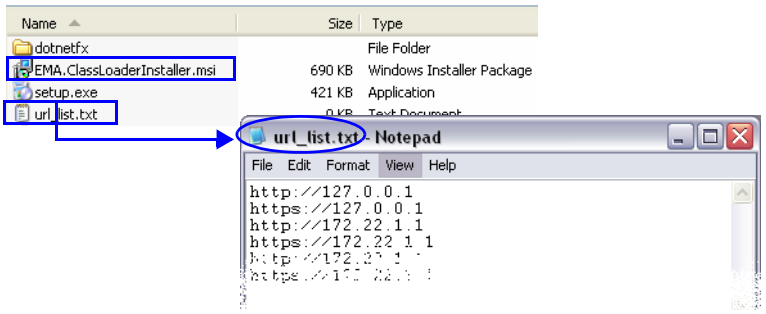
The MSI utility requires the IP addresses of all the RMX systems (both control unit and Shelf Management IP addresses) that each workstation is to connect to.
If the IP address of the any of the target RMXs is changed, the ActiveX components must be reinstalled.

Installing ActiveX

To install ActiveX components on all workstations in the network:

1. Download the MSI file **EMA.ClassLoaderInstaller.msi** from the Polycom Resource Center.
   The MSI file contains installation scripts for both .NET Framework and .NET Security Settings.
2. Create a text file to be used during the installation containing the IP addresses of all the RMX systems (both control unit and Shelf Management IP addresses) that each workstation in the network is to connect to.

   The file must be named **url_list.txt** and must be saved in the same folder as the downloaded MSI file.



3 Install the ActiveX components on all workstations on the network that connect to RMX systems.

The installation is done by the network administrator using a 3rd party network software installation utility and is transparent to all other users.

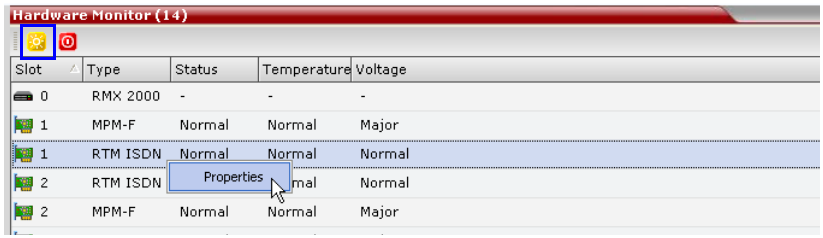
Resetting the RMX

System Reset saves system configuration changes and restarts the system with the latest settings.

To reset the RMX:

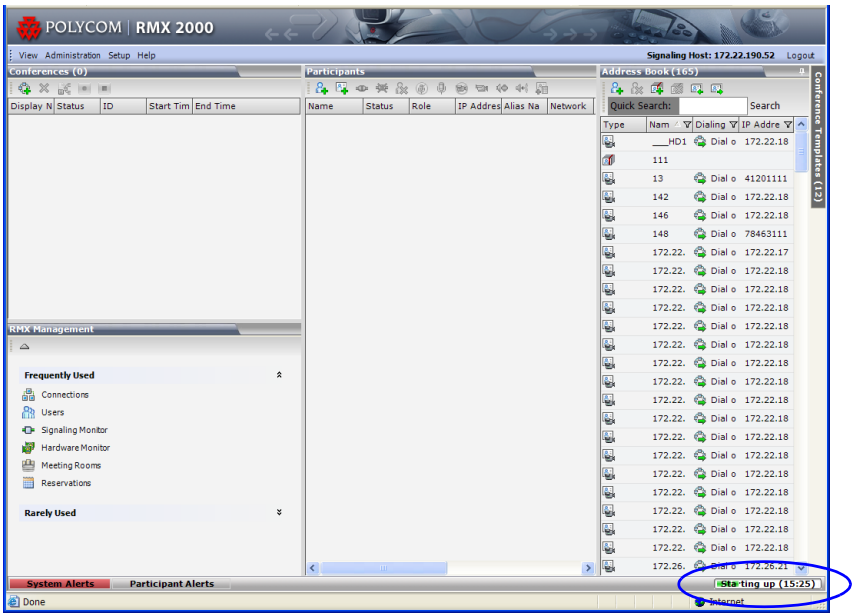
1 In the *RMX Management* pane, click the **Hardware Monitor button**.

The *Hardware Monitor* pane is displayed.



2 Click the **Reset** ( ) button.

When the RMX 2000 is reset, during RMX *Startup* the *Progress Bar* appears at the bottom of the RMX 2000 *Status* pane.



The progress bar displays the amount of time remaining for the reset process to complete:

. The *Startup* progress is also indicated by a green bar moving from left to right.

The duration of the *Startup* depends on the type of activity that preceded the MCU reset. For example: Fast Configuration Wizard, New Version installation, Version Upgrade, Restore Last Configuration etc.

17

Hardware Monitoring

The status and properties of the RMX hardware components can be viewed and monitored in the *Hardware Monitor* list pane.

Viewing the Status of the Hardware Components

The *Hardware Monitor's* status column displays the present status of the hardware components. In addition to the status, temperature and voltage indications are provided for each component.

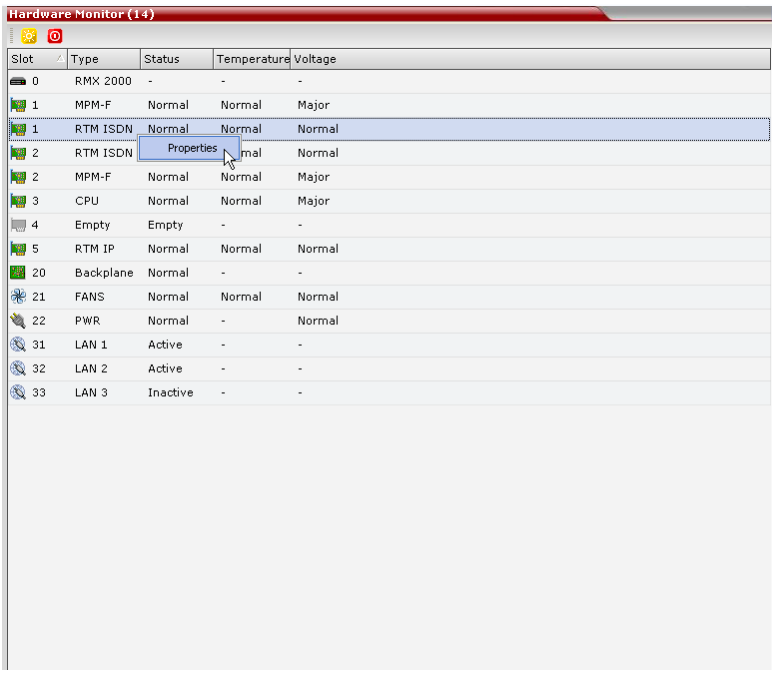
The MCU's Shelf Management Server is what users are connecting to when accessing the *Hardware Monitor* pane. This pane can be accessed in either two ways: through the *RMX Web Client* or the Shelf Management Server. Connection via the Shelf Management Server enables users to access the *Hardware Monitor* even when the connection through the *RMX Web Client* is unavailable. The ability to connect directly via the Shelf Management Server enables users to: enter the *Hardware Monitor* and view the problematic hardware components, reset and restart the MCU and run diagnostics. Running diagnostics and restarting the MCU can only be done via direct connection to the Shelf Management Server. For more information, see "*Diagnostic Mode"* on page **17-13**

When accessing the Shelf Management server, the content displayed will be available in English only.

To view the status of the Hardware Components:

In the *RMX Management* pane, click the **Hardware Monitor** button.

The *Hardware Monitor* pane appears.


Hardware Monitor (14)
Slot
Type
Status
Temperature
Voltage
0 RMX 2000 - - -
1 MPM-F Normal Normal Major
1 RTM ISDN Normal Normal Normal
Properties
2 RTM ISDN Normal Normal
2 MPM-F Normal Normal Major
3 CPU Normal Normal Major
4 Empty Empty - -
5 RTM IP Normal Normal Normal
20 Backplane Normal - -
21 FANS Normal Normal Normal
22 PWR Normal - Normal
31 LAN 1 Active - -
32 LAN 2 Active - -
33 LAN 3 Inactive - -

The *Hardware Monitor* pane displays the following RMX hardware component's status columns:

Table 17-1 *HW Monitor Pane Status Columns*

Field	Description
Slot	Displays an icon according to the HW component type and the slot number. The icon displays the hardware status as follows: • An exclamation point (**!**) indicates errors in the HW component. • Card icon with the reset button ( ) indicates that the HW component is currently resetting. • Card icon with diagnostic tools ( ) indicates that the HW component is in diagnostic mode.

Table 17-1 HW Monitor Pane Status Columns (Continued)

Field	Description
Type	The type of hardware component card.
Status	The current status of the HW component; *Normal, Major, Critical, Resetting, Diagnostics, Active, Inactive or Empty.*
Temperature	Monitors the temperature of the hardware components; Normal, Major and Critical. **Note:** Critical condition invokes a system shut down.
Voltage	The voltage threshold of the hardware component; either *Normal* or *Major*.

HW Monitor Pane Toolbar

The following buttons appear in the toolbar of the Hardware Monitor:

Table 17-2 HW Monitor Pane Toolbar Buttons

Button	Name	Description
	System Reset	Resets and restarts the system. Resetting saves settings and information that you changed in the system, i.e. IP Services, etc...
	System Shut Down	Safely shuts down the system instead of unplugging or manually shutting it down.
	System Start Up	Starts up the system. **Note:** This button is only displayed when connecting directly to the Shelf Management server.
	Diagnostic Mode	Sets the MFA, CPU and Switch (Cards: MPM, CNTL and RTM IP) into diagnostic mode. For more information, see "*Diagnostic Mode"* on page **17-13**. **Note:** This button is only displayed when connecting directly to the Shelf Management server.

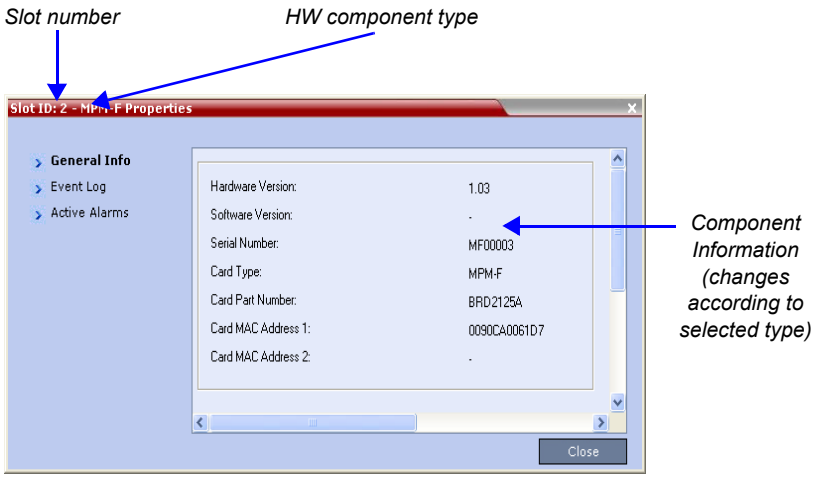
Viewing Hardware Component's Properties

The properties displayed for the hardware components will vary according to the type of component viewed. These component properties can be grouped as follows:

- MCU Properties (RMX)
- Card Properties (MPM F, CPU, RTM IP, RTM ISDN)
- Supporting Hardware Components Properties (Backplane, FANS, LAN)

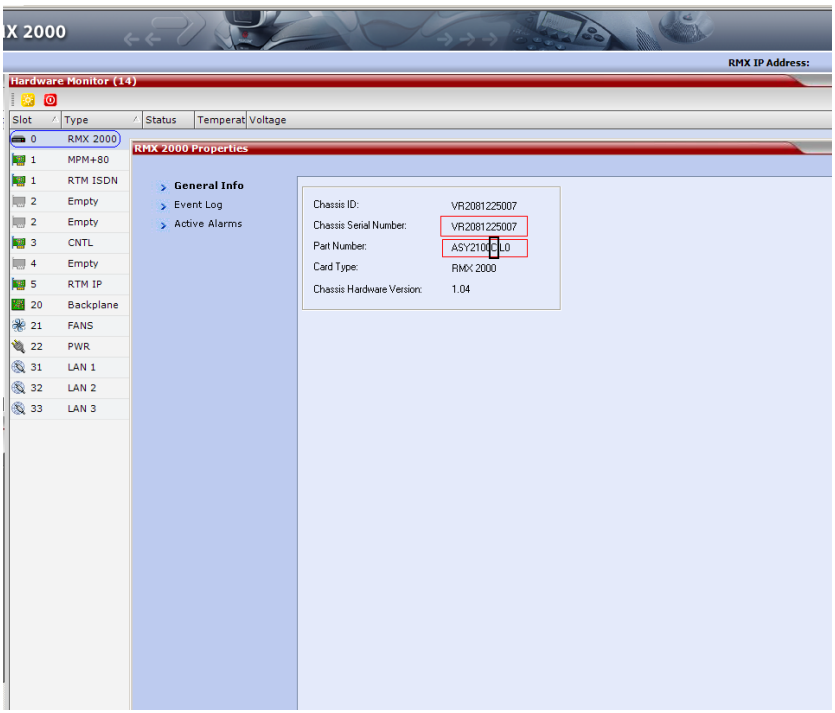
No properties are provided for Power Supply (PWR). For more information, see the *RMX 2000 Hardware Guide*, "*RMX 2000 Specifications*" on page 1-2.

The Hardware Properties dialog box has the following structure:



To view the MCU Properties:

1 In the *Hardware Monitor* pane, either double-click or right-click and select **properties** for *RMX 2000, slot 0*.



The following information is displayed:

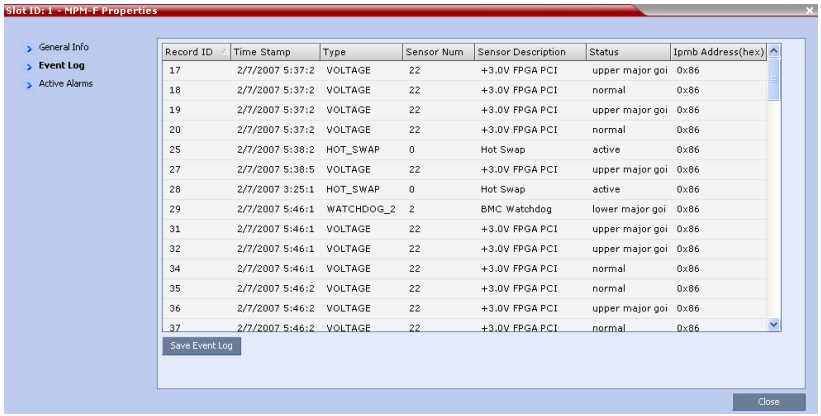
Table 17-3 *MCU Properties - General Info*

Field	Description
Chassis File ID	The ID assigned to the MCU's chassis file.
Chassis Serial Number	The serial number assigned to the MCU's chassis.
Part Number	The chassis part number. The Part Number contains the letter A/B/C/D that represents the chassis type.
Card Type	The name of the hardware product or component, i.e. RMX 2000, Backplane.

Table 17-3 MCU Properties - General Info (Continued)

Field	Description
Chassis HW Version	Indicates the MCU's current chassis hardware version.
Turn SSH	Enables/disables the SSH monitor. This is a secured terminal enabling access to the operating system in order to define Linux commands.

2 Click the *Event Log* tab to view a log of events that were recorded by the system for the *RMX*.


Slot ID: 1 - MPM-F Properties
General Info
Event Log
Active Alarms
Record ID
Time Stamp
Type
Sensor Num
Sensor Description
Status
Ipmb Address(hex)
17 2/7/2007 5:37:2 VOLTAGE 22 +3.0V FPGA PCI upper major goi 0x86
18 2/7/2007 5:37:2 VOLTAGE 22 +3.0V FPGA PCI normal 0x86
19 2/7/2007 5:37:2 VOLTAGE 22 +3.0V FPGA PCI upper major goi 0x86
20 2/7/2007 5:37:2 VOLTAGE 22 +3.0V FPGA PCI normal 0x86
25 2/7/2007 5:38:2 HOT_SWAP 0 Hot Swap active 0x86
27 2/7/2007 5:38:5 VOLTAGE 22 +3.0V FPGA PCI upper major goi 0x86
28 2/7/2007 3:25:1 HOT_SWAP 0 Hot Swap active 0x86
29 2/7/2007 5:46:1 WATCHDOG_2 2 BMC Watchdog lower major goi 0x86
31 2/7/2007 5:46:1 VOLTAGE 22 +3.0V FPGA PCI upper major goi 0x86
32 2/7/2007 5:46:1 VOLTAGE 22 +3.0V FPGA PCI upper major goi 0x86
34 2/7/2007 5:46:1 VOLTAGE 22 +3.0V FPGA PCI normal 0x86
35 2/7/2007 5:46:2 VOLTAGE 22 +3.0V FPGA PCI normal 0x86
36 2/7/2007 5:46:2 VOLTAGE 22 +3.0V FPGA PCI upper major goi 0x86
37 2/7/2007 5:46:2 VOLTAGE 22 +3.0V FPGA PCI normal 0x86
Save Event Log
Close

The logged events can be saved to a *.xls file by clicking the **Save Event Log** button. It is not possible to save individual or multiple selected events; the entire log file must be saved.

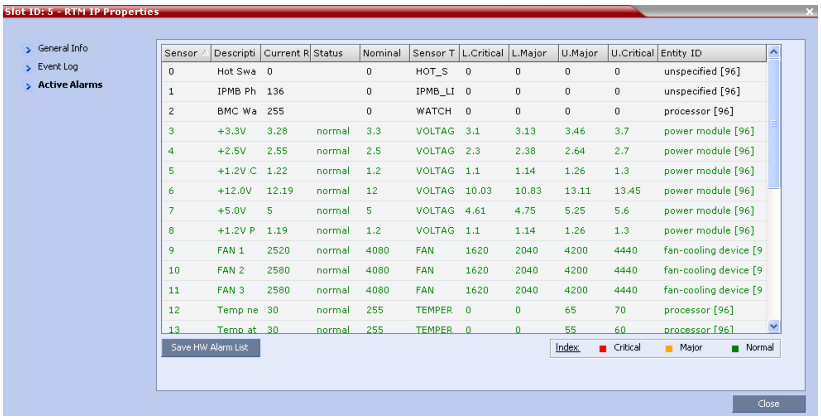
Table 17-4 MCU Properties - Event Log

Column	Description
Record ID	The recorded ID number of the logged event.
Time Stamp	Lists the date and time that the event occurred.
Type	Displays the type of event recorded in the log.
Sensor Number	The number of the LED sensor on the RMX unit.
Sensor Description	Describes which sensor the event is being logged.

Table 17-4 MCU Properties - Event Log (Continued)

Column	Description
Status	The sensor's active status.
Ipmb Address(hex)	Contains all the internal IPMI network addresses on the IPMB bus, i.e. 0x20 (Switch), 0x86 (MFA), etc...

3 Click the *Active Alarms* tab to view alarms related to the *RMX*, i.e. temperatures and main power sensors.


Slot ID: 5 - RTM IP Properties
General Info
Event Log
Active Alarms
Sensor
Descripti
Current R
Status
Nominal
Sensor T
L.Critical
L.Major
U.Major
U.Critical
Entity ID
0 Hot Swa 0 0 HOT_S 0 0 0 0 unspecified [96]
1 IPMB Ph 136 0 IPMB_LI 0 0 0 0 unspecified [96]
2 BMC Wa 255 0 WATCH 0 0 0 0 processor [96]
3 +3.3V 3.28 normal 3.3 VOLTAG 3.1 3.13 3.46 3.7 power module [96]
4 +2.5V 2.55 normal 2.5 VOLTAG 2.3 2.38 2.64 2.7 power module [96]
5 +1.2V C 1.22 normal 1.2 VOLTAG 1.1 1.14 1.26 1.3 power module [96]
6 +12.0V 12.19 normal 12 VOLTAG 10.03 10.83 13.11 13.45 power module [96]
7 +5.0V 5 normal 5 VOLTAG 4.61 4.75 5.25 5.6 power module [96]
8 +1.2V P 1.19 normal 1.2 VOLTAG 1.1 1.14 1.26 1.3 power module [96]
9 FAN 1 2520 normal 4080 FAN 1620 2040 4200 4440 fan-cooling device [9
10 FAN 2 2580 normal 4080 FAN 1620 2040 4200 4440 fan-cooling device [9
11 FAN 3 2580 normal 4080 FAN 1620 2040 4200 4440 fan-cooling device [9
12 Temp ne 30 normal 255 TEMPER 0 0 65 70 processor [96]
13 Temp at 30 normal 255 TEMPER 0 0 55 60 processor [96]
Save HW Alarm List
Index: Critical Major Normal
Close

The *Active Alarms* dialog box displays fields that relate to faults and errors detected on the RMX by sensors. The *Active Alarm*s dialog box is divided into two sections: *HW Alarm List* and *SW Alarm List*.

Each section's alarm list can be saved as a *.xls file by clicking the **Save HW Alarm List** and **Save SW Alarm List** buttons respectively. Each alarm list color codes the severity of the alarm; Critical (RED), Major (ORANGE) and Normal (GREEN).

If you connected to the Hardware Monitoring via the Shelf Management server, the *SW Alarm List* section will not be displayed.

To view the Card Properties:

1 In the *Hardware Monitor* pane, either double-click or right-click and select **properties** for the desired hardware component.

The following information is displayed:

Table 17-5 Card Properties - General Info

Field	Description
HW Version	The hardware component's version number.
SW Version	The version number of the software installed on card.
Serial Number	The hardware component's serial number.
Card Type	Displays the type of card that occupies the slot.
Board Part Number	The part number of the HW component's board.
Board Mac Address 1	Specific hardware address of the component. This address is burnt onto the component and is automatically identified by the system.
Board Mac Address 2	(If applicable) second Mac address.

2 Click the **Event Log** tab to view a log of events that was recorded by the system on the HW component.

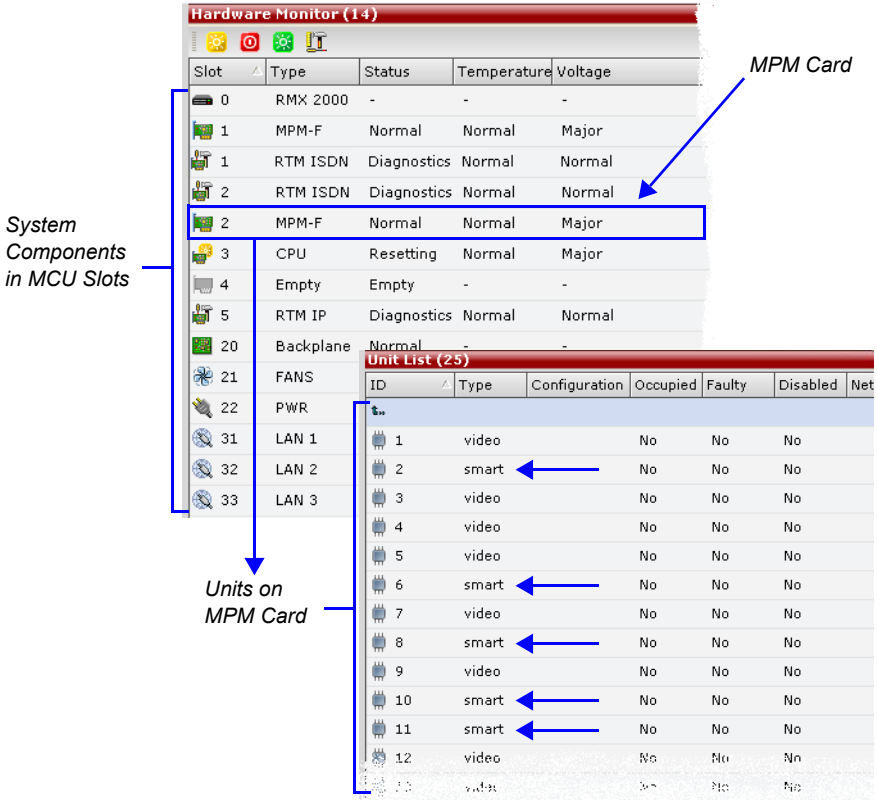
For more information, see "*MCU Properties - Event Log*" on page **17-6**.

3 Click the **Active Alarms** tab to view alarms related to the hardware component, i.e. temperatures and main power sensors.

For more information, see "*Active Alarms*" on page **17-7**.

4 Click **Close** to return to the *HW Monitor* pane.

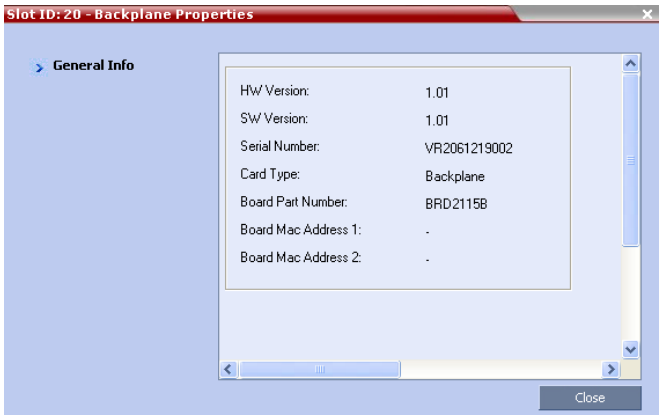
When using the Hardware Monitor to monitor units on MPM cards installed in the RMX's slots, ISDN related DSPs are named *smart*, indicating their additional MUX (Multiplexing) functionality.



To View the Supporting Hardware Components Properties:

1 In the *Hardware Monitor* pane, either double-click or right-click and select properties for the desired supporting hardware component.

The component's properties dialog box will appear with the *General Info* tab displayed.



Backplane Properties:

The RMX unit's backplane properties provides the following information:

Table 17-6 *Backplane Properties- General Info*

Field	Description
HW Version	The Backplane's current hardware version.
SW Version	The Backplane's current software version.
Serial Number	The Backplane's serial number.
Card Type	The name of the hardware component for which information is being displayed, e.g. Backplane.
Board Part Number	The Backplane's part number.
Board Mac Address 1	The Backplane's hardware address.
Board Mac Address 2	(If applicable) second Backplane Mac address.

FAN Properties:

The RMX unit's chassis contains 3 fans that regulate the unit's temperature. If the temperature increases, the fans speed will increase and vice-versa. A "Critical" condition in the fans operation will result in a system shut down.

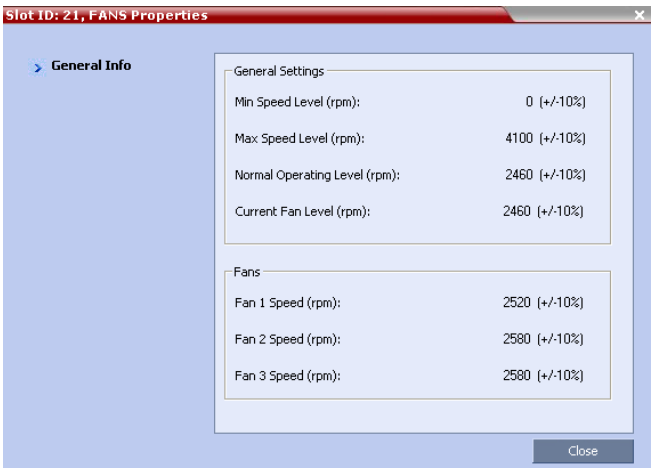
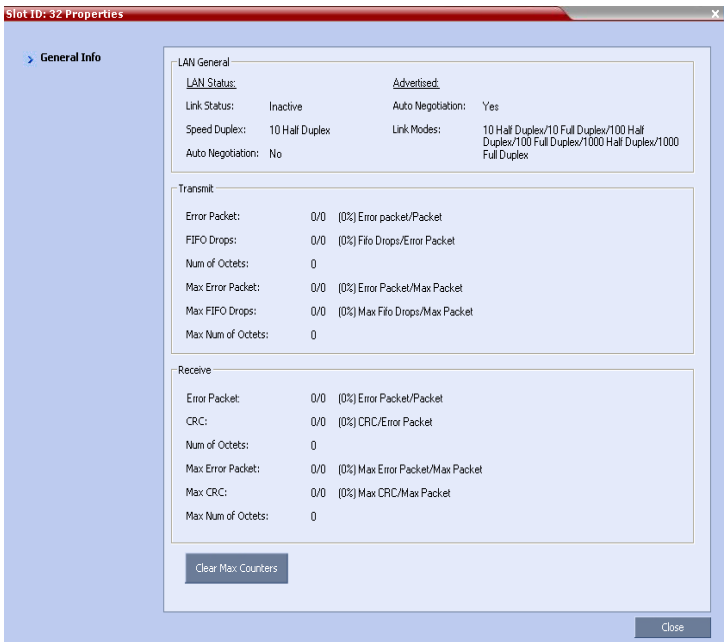

Slot ID: 21, FANS Properties
General Info
General Settings
Min Speed Level (rpm): 0 (+/-10%)
Max Speed Level (rpm): 4100 (+/-10%)
Normal Operating Level (rpm): 2460 (+/-10%)
Current Fan Level (rpm): 2460 (+/-10%)
Fans
Fan 1 Speed (rpm): 2520 (+/-10%)
Fan 2 Speed (rpm): 2580 (+/-10%)
Fan 3 Speed (rpm): 2580 (+/-10%)
Close

Table 17-7 *FANS Properties - General Info*

Field	Description
General Settings	
Min. Speed Level (rpm)	The minimum speed level of the fans.
Max. Speed Level (rpm)	The maximum speed level of the fans.
Normal Operating Level (rpm)	The normal operating level defined for the fans.
Current Fan Level (rpm)	The current operating level of the fans.
Fans	
Fan 1 Speed (rpm)	Present speed of fan 1.
Fan 2 Speed (rpm)	Present speed of fan 2.
Fan 3 Speed (rpm)	Present speed of fan 3.

LAN 0, LAN 1, LAN 2 Properties:

The RMX unit's chassis contains 3 external LAN connectors which register the following information listed below. The information will be refreshed every 8 seconds and also contains a peek detector to log the maximal values, since the last peek values reset.


Slot ID: 32 Properties
General Info
LAN General
LAN Status:
Link Status: Inactive
Speed Duplex: 10 Half Duplex
Auto Negotiation: No
Advertised:
Auto Negotiation: Yes
Link Modes: 10 Half Duplex/10 Full Duplex/100 Half Duplex/100 Full Duplex/1000 Half Duplex/1000 Full Duplex
Transmit
Error Packet: 0/0 (0%) Error packet/Packet
FIFO Drops: 0/0 (0%) Fifo Drops/Error Packet
Num of Octets: 0
Max Error Packet: 0/0 (0%) Error Packet/Max Packet
Max FIFO Drops: 0/0 (0%) Max Fifo Drops/Max Packet
Max Num of Octets: 0
Receive
Error Packet: 0/0 (0%) Error Packet/Packet
CRC: 0/0 (0%) CRC/Error Packet
Num of Octets: 0
Max Error Packet: 0/0 (0%) Max Error Packet/Max Packet
Max CRC: 0/0 (0%) Max CRC/Max Packet
Max Num of Octets: 0
Clear Max Counters
Close

2 Click **Close** to return to the *HW Monitor* pane.

Diagnostic Mode

Diagnostic Mode is a debugging tool for performing hardware diagnostics that detect malfunctions in the hardware component's performance. Diagnostics are performed only for the MFA, CPU and Switch (Cards: MPM, CPU, RTM IP and RTM ISDN). When Diagnostic Mode is initialized, the MCU is reset and upon restarting, the MCU will enter Diagnostic Mode. Entering this mode causes the MCU to terminate all active conferences and prohibits conferences from being established.

Diagnostic Mode is only enabled when connecting directly to the Shelf Management server. To do so, type in the URL address the Shelf Management IP address followed by the system flag: /?DIAG_MODE=true. For example, 172.22.189.51/?DIAG_MODE=true.

You must also be logged in as a SUPPORT user to run diagnostics.

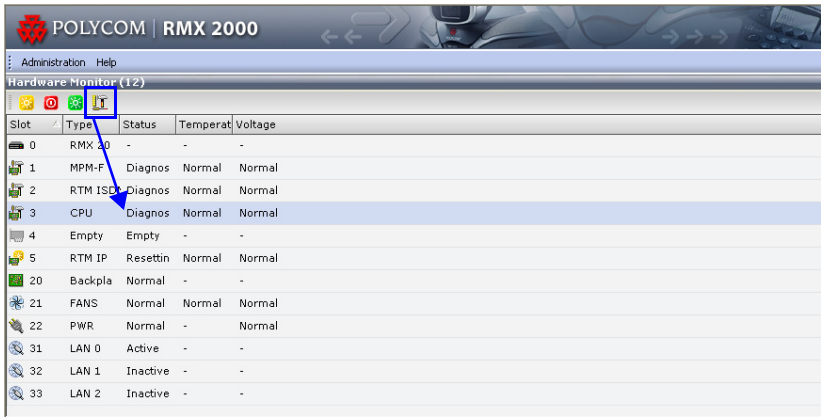
When accessing the Shelf Management server, the content displayed will be available in English only.

Performing Diagnostics

To run Diagnostics on a Hardware Component:

1 In the list pane toolbar, click the **Diagnostic Mode** ( ) button.

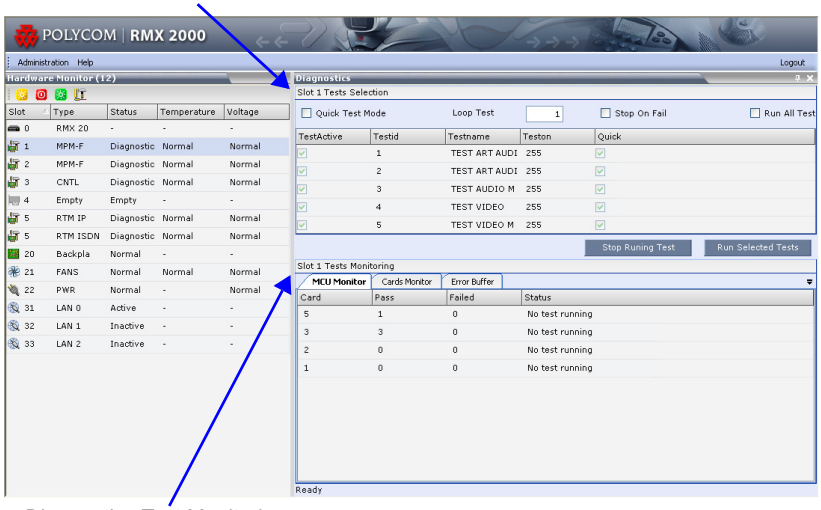
The RTM IP and CNTL components indicate a status of "Diagnostics"; the MPM cards indicate "Resetting". After resetting, the MPM cards will also indicate "Diagnostics" status.


POLYCOM | RMX 2000
Administration Help
Hardware Monitor (12)
Slot Type Status Temperat Voltage
0 RMX 20 - - -
1 MPM-F Diagnos Normal Normal
2 RTM ISD Diagnos Normal Normal
3 CPU Diagnos Normal Normal
4 Empty Empty - -
5 RTM IP Resettin Normal Normal
20 Backpla Normal - -
21 FANS Normal Normal Normal
22 PWR Normal - Normal
31 LAN 0 Active - -
32 LAN 1 Inactive - -
33 LAN 2 Inactive - -

2 Right-click one of the hardware components indicating "Diagnostics" in the status column and select **Diagnostic** from the drop-down menu.

The Diagnostics pane is displayed at the bottom of the screen. Repositioning the pane is enabled by clicking and dragging the pane to the desired location on the screen, i.e. right side bar (displayed below).

Diagnostics Test Selection


POLYCOM | RMX 2000
Administration Help
Logout
Hardware Monitor (12)
Slot Type Status Temperature Voltage
0 RMX 20 - - -
1 MPM-F Diagnostic Normal Normal
2 MPM-F Diagnostic Normal Normal
3 CNTL Diagnostic Normal Normal
4 Empty Empty - -
5 RTM IP Diagnostic Normal Normal
5 RTM ISDN Diagnostic Normal Normal
20 Backpla Normal - -
21 FANS Normal Normal Normal
22 PWR Normal - Normal
31 LAN 0 Active - -
32 LAN 1 Inactive - -
33 LAN 2 Inactive - -
Diagnostics
Slot 1 Tests Selection
Quick Test Mode
Loop Test
1
Stop On Fail
Run All Test
TestActive Testid Testname Teston Quick
1 TEST ART AUDI 255
2 TEST ART AUDI 255
3 TEST AUDIO M 255
4 TEST VIDEO 255
5 TEST VIDEO M 255
Stop Runing Test
Run Selected Tests
Slot 1 Tests Monitoring
MCU Monitor
Cards Monitor
Error Buffer
Card Pass Failed Status
5 1 0 No test running
3 3 0 No test running
2 0 0 No test running
1 0 0 No test running
Ready

Diagnostics Test Monitoring

3 Select the Test(s) to perform in the section labeled *Diagnostics Test Selection* by marking the check boxes in the *TestActive* column.

Each column in the Diagnostics Test Selection is interchangeable. Click, hold and slide a column left or right to the desired position.

The type of tests that can be selected in the *Test Selection* depend on the hardware component. Each component enables different tests.

Additional test parameters can be set before performing the tests, as described below.

Table 17-8 *Tests Selection - Additional Test Parameters*

Parameter	Description
Quick Test Mode	Runs only the tests that are marked **Quick** in the *Quick* column. Particular tests in the system are not as complicated and thus take less time to analyze. These tests are indicated with a check mark in the *Quick* column.
Loop Test	Enter the amount of times the test is to repeat itself in succession.
Stop On Failure	Stops tests upon a failure.
Run All Test	Runs all tests listed in the *TestActive* column for the hardware component.

4 Click the **Run Selected Tests** button.

The selected tests are initialized. This process may take some time. Click **Stop Running Test** to end all the diagnostic tests. The MCU completes the current test running and then stops all remaining tests.

For more information on test results, see "*Diagnostics Monitoring*" on page **17-16**.

5 Repeat procedures 1-6 to run diagnostics for each of the other hardware components.

Diagnostics Monitoring

A hardware component's test status can be viewed in the Diagnostics Test Monitoring section before, during and after tests have been initiated. Test results will only be displayed after tests are completed. The Diagnostic Tests Monitoring section is comprised of three tabs: *MCU Monitor, Cards Monitor* and *Error Buffer*, which are further described below.

MCU Monitor

The MCU Monitor tab lists the status of all the cards that can be tested in Diagnostic Mode. Described below are the columns:

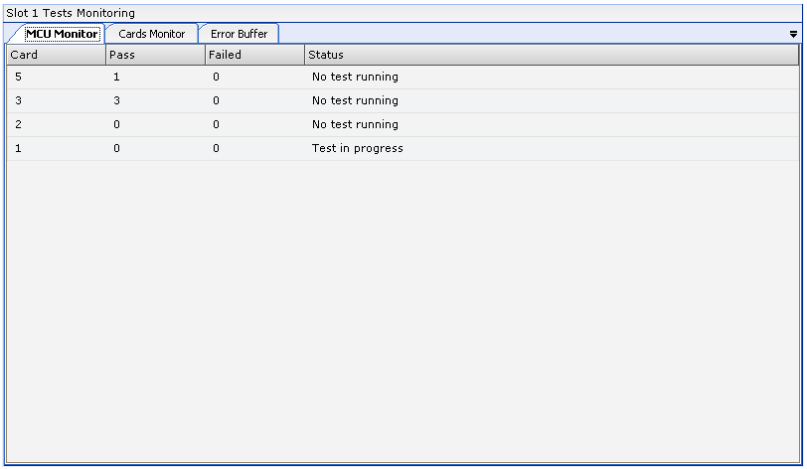


Table 17-9 *Tests Monitoring - MCU Monitor Parameters*

Column	Description
Card	The card's slot number, i.e. 5 - slot where the RTM IP card resides.
Pass	Indicates the number of tests that the card passed successfully.
Fail	Indicates the number of tests that the card failed.
Status	The card's current test status: *No test running* or *Test in progress*.

Cards Monitor

The Cards Monitor tab displays the status of the selected tests being run on the currently viewed card, i.e. slot 5, described below.

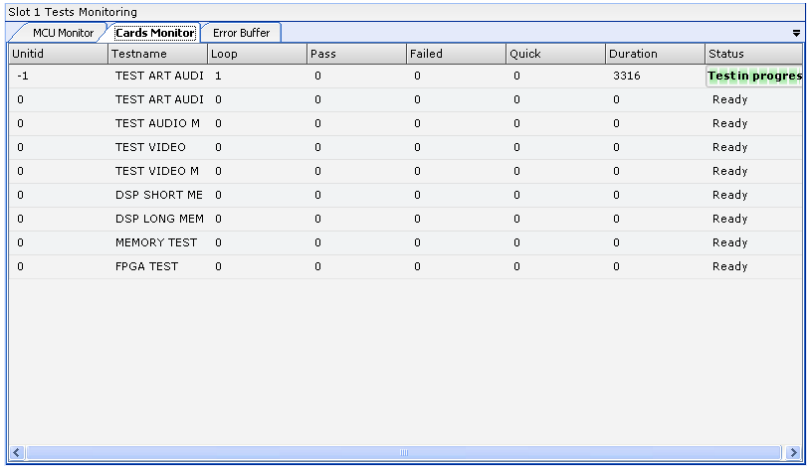

Slot 1 Tests Monitoring
MCU Monitor
Cards Monitor
Error Buffer
Unitid
Testname
Loop
Pass
Failed
Quick
Duration
Status
-1 TEST ART AUDI 1 0 0 0 3316 Test in progres
0 TEST ART AUDI 0 0 0 0 0 Ready
0 TEST AUDIO M 0 0 0 0 0 Ready
0 TEST VIDEO 0 0 0 0 0 Ready
0 TEST VIDEO M 0 0 0 0 0 Ready
0 DSP SHORT ME 0 0 0 0 0 Ready
0 DSP LONG MEM 0 0 0 0 0 Ready
0 MEMORY TEST 0 0 0 0 0 Ready
0 FPGA TEST 0 0 0 0 0 Ready

***Table 17-10** Tests Monitoring - Card Monitor Parameters*

Column	Description
Unitid	The test ID number
Testname	The name of the test
Loop	Indicates the number of times the test will repeat itself in succession.
Pass	Indicates the number of times the test passed successfully.
Failed	Indicates the number of times the test failed.
Quick	Indicates the number of *Quick* tests that have been run on the card.
Duration	The duration of the test (in seconds).
Status	The card's current test status: *Test in Progress* or *Ready*.

Error Buffer

The Error Buffer tab displays the errors encountered during testing of the cards.

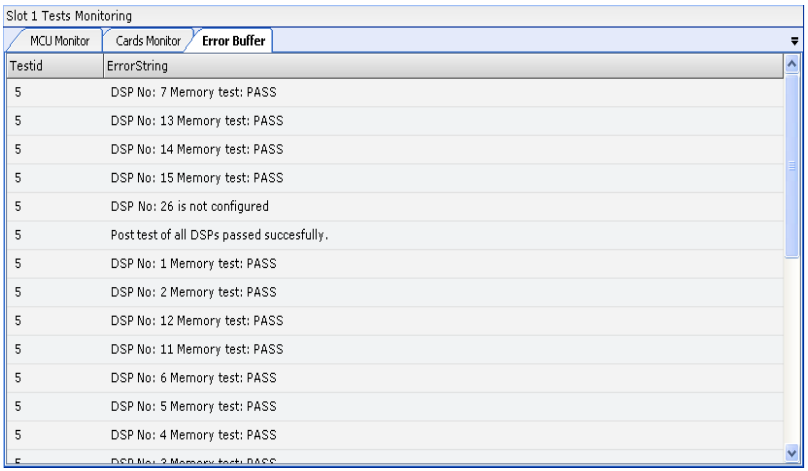

Slot 1 Tests Monitoring
MCU Monitor
Cards Monitor
Error Buffer
Testid
ErrorString
5 DSP No: 7 Memory test: PASS
5 DSP No: 13 Memory test: PASS
5 DSP No: 14 Memory test: PASS
5 DSP No: 15 Memory test: PASS
5 DSP No: 26 is not configured
5 Post test of all DSPs passed succesfully.
5 DSP No: 1 Memory test: PASS
5 DSP No: 2 Memory test: PASS
5 DSP No: 12 Memory test: PASS
5 DSP No: 11 Memory test: PASS
5 DSP No: 6 Memory test: PASS
5 DSP No: 5 Memory test: PASS
5 DSP No: 4 Memory test: PASS

Table 17-11 *Tests Monitoring - Card Monitor Parameters*

Column	Description
Testid	The test ID number.
ErrorString	Indicates the error encountered during testing.

Appendix A

Disconnection Causes

If a participant was unable to connect to a conference or was disconnected from a conference, the **Connection Status** tab in the *Participant Properties* dialog box indicates the call disconnection cause. In some cases, a possible solution may be displayed.

A video participant who is unable to connect the video channels, but is able to connect as an audio only participant, is referred to as a Secondary participant. For Secondary participants, the **Connection Status** tab in the *Participant Properties* dialog box indicates the video disconnection cause. In some cases, a possible solution may be indicated.

The table below lists the call disconnection causes that can be displayed in the Call Disconnection Cause field and provides an explanation of each message

IP Disconnection Causes.

Table A-1 *Call Disconnection Causes*

Disconnection Cause	Description
Disconnected by User	The user disconnected the endpoint from the conference.
Remote device did not open the encryption signaling channel	The endpoint did not open the encryption signaling channel.
Remote devices selected encryption algorithm does not match the local selected encryption algorithm	The encryption algorithm selected by the endpoint does not match the MCU's encryption algorithm.

Table A-1 *Call Disconnection Causes (Continued)*

Disconnection Cause	Description
Resources deficiency	Insufficient resources available.
Call close. Call closed by MCU	The MCU disconnected the call.
H323 call close. No port left for audio	Insufficient audio ports.
H323 call close. No port left for video	The required video ports exceed the number of ports allocated to video in fixed ports.
H323 call close. No port left for FECC	The required data ports exceed the number of ports allocated to data in fixed ports.
H323 call close. No control port left	The required control ports exceed the number of ports allocated to control data in fixed ports.
H323 call close. No port left for videocont	The required video content ports exceed the number of ports allocated to video content in fixed ports.
H323 call closed. Small bandwidth	The gatekeeper allocated insufficient bandwidth to the connection with the endpoint.
H323 call closed. No port left	There are no free ports left in the IP card.
Caller not registered	The calling endpoint is not registered in the gatekeeper.
H323 call closed. ARQ timeout	The endpoint sent an ARQ message to the gatekeeper, but the gatekeeper did not respond before timeout.
H323 call closed. DRQ timeout	The endpoint sent a DRQ message to the gatekeeper, but the gatekeeper did not respond before timeout.
H323 call closed. Alt Gatekeeper failure	An alternate gatekeeper failure occurred.
H323 call closed. Gatekeeper failure	A gatekeeper failure occurred.

Table A-1 *Call Disconnection Causes (Continued)*

Disconnection Cause	Description
H323 call closed. Remote busy	The endpoint was busy. (Applicable only to dial-out)
H323 call closed. Normal	The call ended normally, for example, the endpoint disconnected.
H323 call closed. Remote reject	The endpoint rejected the call.
H323 call closed. Remote unreachable	The gatekeeper could not find the endpoint's address.
H323 call closed. Unknown reason	The reason for the disconnection is unknown, for example, the endpoint disconnected without giving a reason.
H323 call closed. Faulty destination address	Incorrect address format.
H323 call closed. Small bandwidth	The gatekeeper allocated insufficient bandwidth to the connection with the endpoint.
H323 call closed. Gatekeeper reject ARQ	The gatekeeper rejected the endpoint's ARQ.
H323 call closed. No port left	There are no ports left in the IP card.
H323 call closed. Gatekeeper DRQ	The gatekeeper sent a DRQ.
H323 call closed. No destination IP address	For internal use.
H323 call. Call failed prior or during the capabilities negotiation stage	The endpoint did not send its capabilities to the gatekeeper.
H323 call closed. Audio channels didn't open before timeout	The endpoint did not open the audio channel.
H323 call closed. Remote sent bad capability	There was a problem in the capabilities sent by the endpoint.

Table A-1 *Call Disconnection Causes (Continued)*

Disconnection Cause	Description
H323 call closed. Local capability wasn't accepted by remote	The endpoint did not accept the capabilities sent by the gatekeeper.
H323 failure	Internal error occurred.
H323 call closed. Remote stop responding	The endpoint stopped responding.
H323 call closed. Master slave problem	A People + Content cascading failure occurred.
SIP bad name	The conference name is incompatible with SIP standards.
SIP bad status	A general IP card error occurred.
SIP busy everywhere	The participant's endpoints were contacted successfully, but the participant is busy and does not wish to take the call at this time.
SIP busy here	The participant's endpoint was contacted successfully, but the participant is currently not willing or able to take additional calls.
SIP capabilities don't match	The remote device capabilities are not compatible with the conference settings.
SIP card rejected channels	The IP card could not open the media channels.
SIP client error 400	The endpoint sent a SIP Client Error 400 (Bad Request) response. The request could not be understood due to malformed syntax.
SIP client error 402	The endpoint sent a SIP Client Error 402 (Payment Required) response.
SIP client error 405	The endpoint sent a SIP Client Error 405 (Method Not Allowed) response. The method specified in the Request-Line is understood, but not allowed for the address identified by the Request-URI.

Table A-1 *Call Disconnection Causes (Continued)*

Disconnection Cause	Description
SIP client error 406	The endpoint sent a SIP Client Error 406 (Not Acceptable) resources. The remote endpoint cannot accept the call because it does not have the necessary responses. The resource identified by the request is only capable of generating response entities that have content characteristics not acceptable according to the Accept header field sent in the request.
SIP client error 407	The endpoint sent a SIP Client Error 407 (Proxy Authentication Required) response. The client must first authenticate itself with the proxy.
SIP client error 409	The endpoint sent a SIP Client Error 409 (Conflict) response. The request could not be completed due to a conflict with the current state of the resource.
SIP client error 411	The endpoint sent a SIP Client Error 411 (Length Required) response. The server refuses to accept the request without a defined Content Length.
SIP client error 413	The endpoint sent a SIP Client Error 413 (Request Entity Too Large) response. The server is refusing to process a request because the request entity is larger than the server is willing or able to process.
SIP client error 414	The endpoint sent a SIP Client Error 414 (Request-URI Too Long) response. The server is refusing to service the request because the Request-URI is longer than the server is willing to interpret.
SIP client error 420	The endpoint sent a SIP Client Error 420 (Bad Extension) response. The server did not understand the protocol extension specified in a Require header field.

Table A-1 *Call Disconnection Causes (Continued)*

Disconnection Cause	Description
SIP client error 481	The endpoint sent a SIP Client Error 481 (Call/ Transaction Does Not Exist) response.
SIP client error 482	The endpoint sent a SIP Client Error 482 (Loop Detected) response.
SIP client error 483	The endpoint sent a SIP Client Error 483 (Too Many Hops) response.
SIP client error 484	The endpoint sent a SIP Client Error 484 (Address Incomplete) response. The server received a request with a To address or Request-URI that was incomplete.
SIP client error 485	The endpoint sent a SIP Client Error 485 (Ambiguous) response. The address provided in the request (Request-URI) was ambiguous.
SIP client error 488	The endpoint sent a SIP Client Error 488 (Not Acceptable Here) response.
SIP forbidden	The SIP server rejected the request. The server understood the request, but is refusing to fulfill it.
SIP global failure 603	A SIP Global Failure 603 (Decline) response was returned. The participant's endpoint was successfully contacted, but the participant explicitly does not wish to or cannot participate.
SIP global failure 604	A SIP Global Failure 604 (Does Not Exist Anywhere) response was returned. The server has authoritative information that the user indicated in the Request-URI does not exist anywhere.
SIP global failure 606	A SIP Global Failure 606 (Not Acceptable) response was returned.
SIP gone	The requested resource is no longer available at the Server and no forwarding address is known.

Table A-1 *Call Disconnection Causes (Continued)*

Disconnection Cause	Description
SIP moved permanently	The endpoint moved permanently. The user can no longer be found at the address in the Request-URI.
SIP moved temporarily	The remote endpoint moved temporarily.
SIP not found	The endpoint was not found. The server has definitive information that the user does not exist at the domain specified in the Request-URI.
SIP redirection 300	A SIP Redirection 300 (Multiple Choices) response was returned.
SIP redirection 305	A SIP Redirection 305 (Use Proxy) response was returned. The requested resource MUST be accessed through the proxy given by the Contact field.
SIP redirection 380	A SIP Redirection 380 (Alternative Service) response was returned. The call was not successful, but alternative services are possible.
SIP remote cancelled call	The endpoint canceled the call.
SIP remote closed call	The endpoint ended the call.
SIP remote stopped responding	The endpoint is not responding.
SIP remote unreachable	The endpoint could not be reached.
SIP request terminated	The endpoint terminated the request. The request was terminated by a BYE or CANCEL request.
SIP request timeout	The request was timed out.
SIP server error 500	The SIP server sent a SIP Server Error 500 (Server Internal Error) response. The server encountered an unexpected condition that prevented it from fulfilling the request.

Table A-1 *Call Disconnection Causes (Continued)*

Disconnection Cause	Description
SIP server error 501	The SIP server sent a SIP Server Error 501 (Not Implemented) response. The server does not support the functionality required to fulfill the request.
SIP server error 502	The SIP server sent a SIP Server Error 502 (Bad Gateway) response. The server, while acting as a gateway or proxy, received an invalid response from the downstream server it accessed in attempting to fulfill the request.
SIP server error 503	The SIP server sent a SIP Server Error 503 (Service Unavailable) response. The server is temporarily unable to process the request due to a temporary overloading or maintenance of the server.
SIP server error 504	The SIP server sent a SIP Server Error 504 (Server Time-out) response. The server did not receive a timely response from an external server it accessed in attempting to process the request.
SIP server error 505	The SIP server sent a SIP Server Error 505 (Version Not Supported) response. The server does not support, or refuses to support, the SIP protocol version that was used in the request.
SIP temporarily not available	The participant's endpoint was contacted successfully but the participant is currently unavailable (e.g., not logged in or logged in such a manner as to preclude communication with the participant).
SIP remote device did not respond in the given time frame	The endpoint did not respond in the given time frame.
SIP trans error TCP Invite	A SIP Invite was sent via TCP, but the endpoint was not found.

Table A-1 *Call Disconnection Causes (Continued)*

Disconnection Cause	Description
SIP transport error	Unable to initiate connection with the endpoint.
SIP unauthorized	The request requires user authentication.
SIP unsupported media type	The server is refusing to service the request because the message body of the request is in a format not supported by the requested resource for the requested method.

ISDN Disconnection Causes

Table A-2 *ISDN Disconnection Causes*

Disconnection Cause		
Number	**Summary**	**Description**
1	*Unallocated (unassigned number)*	No route to the number exists in the ISDN network or the number was not found in the routing table. • Ensure that the number appears in the routing table. • Ensure that it is a valid number and that correct digits were dialed.
2	*No route to specified transit network (national use)*	The route specified (transit network) between the two networks does not exist.
3	*No route to destination*	No physical route to the destination number exists although the dialed number is in the routing plan. • The PRI D-Channel is malfunctioning. • Incorrect connection of the span or WAN.
4	*Send special information tone*	Return the special information tone to the calling party indicating that the called user cannot be reached.
5	*Misdialed trunk prefix (national use)*	A trunk prefix has erroneously been included in the called user number.
6	*Channel Unacceptable*	The sending entity in the call does not accept the channel most recently identified.
7	*Call awarded and being delivered in an Established channel*	The incoming call is being connected to a channel previously established for similar calls.

Table A-2 *ISDN Disconnection Causes (Continued)*

Disconnection Cause		
Number	**Summary**	**Description**
8	*Pre-Emption*	The call has been pre-empted.
9	*Pre-Emption – Circuit reserved for reuse*	Call is being cleared in response to user request.
16	*Normal Call Clearing*	Call cleared normally because user hung up.
17	*User Busy*	Dialed number is busy.
18	*No User Responding*	The called user has not answered the call.
19	*No Answer from User (User Alerted)*	Called user has received call alert, but has not responded within a prescribed period of time. Internal network timers may initiate this disconnection.
20	*Subscriber Absent*	User is temporarily absent from the network - as when a mobile user logs off.
21	*Call Rejected*	Called number is either busy or has compatibility issues. Supplementary service constraints in the network may also initiate the disconnection.
22	*Number Changed*	Same as Cause 1. The diagnostic field contains the new called user number. Cause 1 is used if the network does not support this cause value.
26	*Non-Selected User Clearing*	The incoming call has not been assigned to the user.
27	*Destination Out-of-Order*	Messages cannot be sent to the destination number because the span may not be active.
28	*Invalid Number Format (address incomplete)*	The Type of Number (TON) is incorrect or the number is incomplete. Network, Unknown and National numbers have different formats.

Table A-2 *ISDN Disconnection Causes (Continued)*

Disconnection Cause		
Number	**Summary**	**Description**
29	*Facility Rejected*	User requested supplementary service which cannot be provided by the network.
30	*Response to STATUS ENQUIRY*	A STATUS message has been received in response to a prior STATUS ENQUIRY.
31	*Normal, Unspecified*	A normal, unspecified disconnection has occurred.
34	*No Circuit/ Channel Available*	No B-Channels are available for the call.
38	*Network Out-of-Order*	Network is out-of-order because due to a major malfunction.
39	*Permanent Frame Mode Connection Out-of-Service*	A permanent frame mode connection is out-of-service. This cause is part of a STATUS message.
40	*Permanent Frame Mode Connection Operational*	A permanent frame mode connection is operational. This cause is part of a STATUS message.
41	*Temporary Failure*	Minor network malfunction. Initiate call again.
42	*Switching Equipment Congestion*	High traffic has congested the switching equipment. Cause 43 is included.
43	*Access Information Discarded*	Access Information elements exceed maximum length and have been discarded. Included with Cause 42.
44	*Requested Circuit/Channel not Available*	The requested circuit or channel is not available. Alternative circuits or channels are not acceptable.

Table A-2 *ISDN Disconnection Causes (Continued)*

Disconnection Cause		
Number	**Summary**	**Description**
47	*Resource Unavailable, Unspecified*	The resource is unavailable. No other disconnection cause applies.
49	*Quality of Service Not Available*	Quality of Service, as defined in Recommendation X.213, cannot be provided.
50	*Requested Facility Not Subscribed*	A supplementary service has been requested that the user is not authorized to use.
53	*Outgoing Calls Barred Within Closed User Group (CUG)*	Outgoing calls are not permitted for this member of the CUG.
55	*Incoming Calls Barred within CUG*	Incoming calls are not permitted for this member of the CUG.
57	*Bearer Capability Not Authorized*	A bearer capability has been requested that the user is not authorized to use.
58	*Bearer Capability Not Presently Available*	A bearer capability has been requested that the user is not presently available.
62	*Inconsistency in Designated Outgoing Access Information and Subscriber Class*	Outgoing Access and Subscriber Class information is inconsistent

Table A-2 *ISDN Disconnection Causes (Continued)*

Disconnection Cause		
Number	**Summary**	**Description**
63	*Service or Option Not Available, Unspecified*	The service or option is unavailable. No other disconnection cause applies.
65	*Bearer Capability Not Implemented*	The requested bearer capability is not supported.
66	*Channel Type Not Implemented*	The requested channel type is not supported.
69	*Requested Facility Not Implemented*	The requested supplementary service is not supported.
70	*Only Restricted Digital Information Bearer Capability is Available (national use)*	Unrestricted (64kb) bearer service has been requested but is not supported by the equipment sending this cause.
79	*Service or Option Not Implemented, Unspecified*	An unsupported service or unimplemented option has been requested. No other disconnection cause applies.
81	*Invalid Call Reference Value*	A message has been received which contains a call reference which is currently unassigned or not in use on the user-network interface.
82	*Identified Channel Does Not Exist*	A request has been received to use a channel which is currently inactive or does not exist.

Table A-2 *ISDN Disconnection Causes (Continued)*

Disconnection Cause		
Number	**Summary**	**Description**
83	*A Suspended Call Exists, but This Call Identity Does Not Exist*	A RESUME message cannot be executed by the network as a result of an unknown call identity.
84	*Call Identity in Use*	A SUSPEND message has been received with a call identity sequence that is already in use.
85	*No Call Suspended*	A RESUME message cannot be executed by the network as a result of no call suspended.
86	*Call Having the Requested Call Identity Has Been Cleared*	A RESUME message cannot be executed by the network as a result of the call having been cleared while suspended.
87	*User Not Member of CUG*	A CUG member was called by a user that is not a member of the CUG or a CUG call was made to a non CUG member.
88	*Incompatible Destination*	User-to-user compatibility checking procedures in a point-to-point data link have determined that an incompatibility exists between Bearer capabilities.
90	*Non-Existent CUG*	CUG does not exist.
91	*Invalid Transit Network Selection (national use)*	The transit network selection is of an incorrect format. No route (transit network) exists between the two networks.
95	*Invalid Message, Unspecified*	Invalid message received. No other disconnection cause applies.
96	*Mandatory Information Element is Missing*	A message was received with an information element missing.

Table A-2 *ISDN Disconnection Causes (Continued)*

Disconnection Cause		
Number	**Summary**	**Description**
97	*Message Type Non-Existent or Not Implemented*	A message was received that is of a type that is not defined or of a type that is defined but not implemented.
98	*Message is Not Compatible with the Call State, or the Message Type is Non-Existent or Not Implemented*	An unexpected message or unrecognized message incompatible with the call state has been received
99	*An Information Element or Parameter Does Not Exist or is Not Implemented*	A message was received containing elements or parameters that are not defined or of a type that is defined but not implemented.
100	*Invalid Information Element Contents*	A message other than SETUP, DISCONNECT, RELEASE, or RELEASE COMPLETE has been received which has one or more mandatory information elements containing invalid content.
101	*The Message is Not Compatible with the Call State*	A STATUS message indicating any call state except the Null state has been received while in the Null state.
102	*Recovery on Timer Expired*	An error handling procedure timer has expired.
103	*Parameter Non-Existent or Not Implemented – Passed On (national use)*	A message was received containing parameters that are not defined or of a type that is defined but not implemented.

Table A-2 *ISDN Disconnection Causes (Continued)*

Disconnection Cause		
Number	**Summary**	**Description**
110	*Message with Unrecognized Parameter Discarded*	A message was discarded because it contained a parameter that was not recognized.
111	*Protocol Error, Unspecified*	A protocol error has occurred. No other disconnection cause applies.
127	*Interworking, Unspecified*	An interworking call has ended.

Appendix B

Alarms and Faults

Alarms

Table B-1 *Alarms*

Alarm Code	Alarm Description
A new activation key was loaded. Reset the system.	A new activation key was loaded: Reset the MCU.
A new version was installed. Reset the system.	A new version was installed: Reset the MCU.
A private version is loaded	A private version is loaded: [private description].
Action redirection failure	Possible explanations: • Action redirection failure. • Action redirection map incomplete.
Automatic reset is unavailable in Safe Mode	The system switches to safe mode if many resets occur during startup. To prevent additional resets, and allow the system to complete the startup process the automatic system resets are blocked.
Card failure	Possible reasons for the card failure: • Resetting Card • Resetting component • Unknown shelf error • Unknown card error
Card not found	This occurs when: the system does not receive an indication about the card (since it does not exist…) usually when the card was removed from the MCU and the system did not have a chance to recalculate it resources.

Table B-1 *Alarms (Continued)*

Alarm Code	Alarm Description
Card not responding	Possible reasons for the card not responding: • No connection with MPM card. • No connection with the Switch.
Central signaling component failure	Possible explanations: • Central signaling component failure; unit type: [NonComponent\CSMngnt\CSH323\CSSIP] • Central signaling component failure; unit type: (invalid: [NonComponent\CSMngnt\CSH323\CSSIP]) • Central signaling component failure - Invalid failure type. Unit id: [id], Type: [NonComponent\CSMngnt\CSH323\CSSIP], Status: [Ok\Failed\Recovered] • Central signaling component failure - Invalid failure type
Central Signaling indicating Faulty status	Central signaling failure detected in IP Network Service.
Central Signaling indicating Recovery status	
Central Signaling startup failure	
Configuration of external database did not complete.	
Could not complete MPM Card startup procedure	Possible explanations: • Unit loading confirmation was not received. • No Media IP for this card. • Media IP Configuration confirmation was not received. • Unspecified problem.
Could not complete RTM ISDN Card startup procedure	
CPU IPMC software was not updated.	
D channel cannot be established	

Table B-1 Alarms (Continued)

Alarm Code	Alarm Description
DEBUG mode enabled	Possible explanations: • System is running in DEBUG mode. • System DEBUG mode initiated.
DEBUG mode flags in use	System is using DEBUG CFG flags.
DMA not supported by IDE device	Possible explanations: • DMA (direct memory access) not supported by IDE device: Incompatible flash card / hard disk being used. • Flash card / hard drive are not properly connected to the board / one of the IDE channels is disconnected. • DMA was manually disabled for testing.
DNS configuration error	
DNS not configured in IP Network Service	
Error in external database certificate	
Error reading MCU time	Failed to read MCU time configuration file ([status]).
External NTP servers failure	
Failed to access DNS server	Failed to access DNS server.
Failed to configure the Media card IP address	Possible reasons for the failure: • Failure type: [OK Or Not supported. • Does not exist Or IP failure. • Duplicate IP Or DHCP failure. • VLAN failure Or Invalid: [status_Number].
Failed to configure the Users list in Linux	External NTP server failure: NTP server failure: [server0_ip], [server1_ip], [server2_ipStr].
Failed to connect to application server	Possible reasons for the failure: • Failed to connect to application server: • Failed to establish connection to server, url = [url].

Table B-1 Alarms (Continued)

Alarm Code	Alarm Description
Failed to connect to recording device	The MCU could not connect to any of the defined NTP server for synchronization due to the remote server error.
Failed to connect to SIP registrar	Cannot establish connection with SIP registrar.
Failed to create Default Profile	Possible reasons for the failure: • Failed to validate the Default Profile. • Failed to add the Default Profile.
Failed to initialize the file system	Possible reasons for the failure: • Failed to initialize the file system. • Failed to initialize the file system and create the CDR index.
Failed to mount Card folder	Failed to mount card folder.
Failed to open Apache server configuration file	Failed to open Apache configuration file.
Failed to open Users list file	
Failed to register with DNS server	
Failed to save Apache server configuration file	Failed to save Apache configuration file.
Failure in initialization of SNMP agent.	
Fallback version is being used	Fallback version is being used. Restore current version. Version being used: [running version]; Current version: [current version].
File error	Possible reasons for the file error: • XML file does not exist [file name]; Error no: [error number]. • Not authorized to open XML file [file name]; Error no: [error number]. • Unknown problem in opening XML file [file name]; Error no: [error number]. • Failed to parse XML file [file name].

Table B-1 *Alarms (Continued)*

Alarm Code	Alarm Description
File system scan failure	File system scan failure: Failed to scan [file system path].
File system space shortage	File system space shortage: Out of file system space in [file system path]; Free space: [free space percentage]% ([free space] Blocks) - Minimum free space required: [minimum free space percentage]% ([minimum free space] Blocks).
Gatekeeper failure	Possible reasons for the Gatekeeper failure: • Failed to register to alternate Gatekeeper. • Gatekeeper discovery state. - Check GK IP address (GUI, ping) • Gatekeeper DNS Host name not found. • Gatekeeper Registration Timeout. • Gatekeeper rejected GRQ due to invalid revision. • Gatekeeper rejected GRQ due to resource unavailability. • Gatekeeper rejected GRQ due to Terminal Exclusion. • Gatekeeper rejected GRQ due to unsupported feature. • Gatekeeper rejected GRQ. Reason 18. • Gatekeeper rejected RRQ due to Discovery Required. • Gatekeeper rejected RRQ due to duplicate alias. - Check duplicate in aliases or in prefixes • Gatekeeper rejected RRQ due to Generic Data. • Gatekeeper rejected RRQ due to invalid alias. • Gatekeeper rejected RRQ due to invalid call signaling address. • Gatekeeper rejected RRQ due to invalid endpoint ID. • Gatekeeper rejected RRQ due to invalid RAS address. • Gatekeeper rejected RRQ due to invalid revision. • Gatekeeper rejected RRQ due to invalid state.

Table B-1 *Alarms (Continued)*

Alarm Code	Alarm Description
Gatekeeper failure (cont.)	• Gatekeeper rejected RRQ due to invalid terminal alias. • Gatekeeper rejected RRQ due to resource unavailability. • Gatekeeper rejected RRQ due to Security Denial. • Gatekeeper rejected RRQ due to terminal type. • Gatekeeper rejected RRQ due to unsupported Additive Registration. • Gatekeeper rejected RRQ due to unsupported feature. • Gatekeeper rejected RRQ due to unsupported QOS transport. • Gatekeeper rejected RRQ due to unsupported transport. • Gatekeeper rejected RRQ. Full registration required. • Gatekeeper rejected RRQ. Reason 18. • Gatekeeper Unregistration State. • Registration succeeded.
Hard disk error	Hard disk not responding.
High CPU utilization	
High system CPU usage	High system CPU usage: System CPU usage is approaching limit.
Incorrect Ethernet Settings	Incorrect Ethernet Settings: Ethernet should be set to 100 Full Duplex, Auto Negotiation - off.
Insufficient resources	Insufficient resources.
Insufficient UDP Ports	

Table B-1 Alarms (Continued)

Alarm Code	Alarm Description
Internal MCU reset	Possible explanations: • McmsDaemon reset due to policy decision: [Process failed [abs crash counter: crash counter]: process name]. • McmsDaemon reset due to policy decision: [Process failed [abs crash counter: crash counter]: process name]; Cannot reset while system is in DEBUG mode. • Power down signal was detected. • [CS Component Failure; unit type: [NonComponent\CSMngnt\CSH323\CSSIP]\CS Component Failure; unit type: (invalid: [unit type])\No connection with CS]; Cannot reset while system is in DEBUG mode. • Reset cause unknown: [reset source]\Restore Factory Defaults - [mcu restore name]\CM_Loaded indication repeated; boardId: [boardId]\reset from Cards process - simulation\No connection with MPM; board Id:[boardId]\SmMfaFailure - boardId: [boardId]. Status: [status], problem bitmask: [problemBitMask]\MPM failure, boardId: [slotId]\Switch failure\No connection with Switch.
Internal System configuration during startup	System configuration during startup.
Invalid date and time	Invalid date and time: MCU year ([year]) must be 2000 or later.
Invalid MCU Version	MCU Version: [Major.Minor.release.internal].
Invalid System Configuration	
IP addresses of Signaling Host and Control Unit are the same	
IP Network Service configuration modified	IP Network Service was modified. Reset the MCU.
IP Network Service deleted	IP Network Service was deleted. Reset the MCU.

Table B-1 *Alarms (Continued)*

Alarm Code	Alarm Description
IP Network Service not found	Possible explanations: • IP Service not found in the Network Services list. • m_StatusRead IpServiceList.
ISDN/PSTN Network Services configuration changed	
License not found	
Low Processing Memory	Low Processing Memory: Process is approaching memory utilization limit: [Memory Utilization Percent]
Low system Memory	Low system Memory: The system exceeded 80% of memory usage.
Management Network not configured	
MCU is not configured for AVF gatekeeper mode	
MCU reset	The MCU was reset automatically or by the user. MCU reset: Reset cause: [reset source].
MCU Reset to enable Diagnostics mode	
Missing Central Signaling configuration	
MPL startup failure. Authentication not received.	
MPL startup failure. Management Network configuration not received.	
Music file error	The music file played during the connection to the conference cannot be accessed.
No clock source	The system could not use any of the connected ISDN spans as clock source

Table B-1 Alarms (Continued)

Alarm Code	Alarm Description
No default ISDN/PSTN Network Service defined in ISDN/PSTN Network Services list	
No default IVR Service in IVR Services list	No default IVR Service in IVR Services list: Ensure that one conference IVR Service and one EQ IVR Service are set as default.
No IP Network Services defined	IP Network Service parameters missing.
No ISDN/PSTN Network Services defined	No ISDN/PSTN Network Services were defined or no default ISDN/PSTN Network was defined.
No License for ISDN/PSTN. Please activate the RTM ISDN card through Polycom website	
No response from Central Signaling	No connection with central signaling.
No response from RTM ISDN card	
No usable unit for audio controller;	
NTP synchronization failure	The system failed to synchronize the MCU clock with the NTP clock
Polycom default User exists. For security reasons, it is recommended to delete this User and create your own User.	
Port configuration was modified	
Power off	
Process idle	Process idle: Process did not finish before deadline.
Process terminated	Process terminated: [Process name] terminated.
Product activation failure	
Recording device has disconnected unexpectedly	

Table B-1 *Alarms (Continued)*

Alarm Code	Alarm Description
Red Alarm	When a certain timeout will be reached (after startup), MCMS will go over the configured Spans. A configured Span that is related to nonexistent card – will produce a 'RED_ALARM' Alert. Similarly on HotSwap: if an RTM card (or an MPM that has an RTM extension) is removed, MCMS will go over the configured Spans. A configured Span that is related to the removed card – will produce a 'RED_ALARM' Alert.
Resource process did not receive the Meeting Room list during startup.	Without the Meeting Rooms list, the system cannot allocate the appropriate dial numbers, Conference ID etc. and therefore cannot run conferences
Resource process failed to request the Meeting Room list during startup.	Without the Meeting Rooms list, the system cannot allocate the appropriate dial numbers, Conference ID etc. and therefore cannot run conferences
Restoring Factory Defaults. Default system settings will be restored once Reset is completed	Default system settings will be restored once Reset is completed.
RTM ISDN card not found	RTM ISDN card is missing.
RTM ISDN card startup procedure error	The RTM ISDN card cannot complete its startup procedure (usually after system reset)
Secured SIP communication failed	
Security mode failed. Certificate not yet valid.	
Security mode failed. Certificate has expired.	
Security mode failed. Certificate host name does not match the RMX host name.	
Security mode failed. Certificate is about to expire.	
Security mode failed. Error in certificate file.	

Table B-1 Alarms (Continued)

Alarm Code	Alarm Description
Single clock source	No Backup clock could be established as only one span is connected to the system or, there is a synchronization failure with another span. This alarm can be cancelled by adding the appropriate flag is the system configuration.
SIP registrations limit reached	SIP registrations limit reached.
SIP TLS: Failed to load or verify certificate files	This alarm indicates that the certificate files required for SIP TLS could not be loaded to the RMX. Possible causes are: • Incorrect certificate file name. Only files with the following names can be loaded to the system: rootCA.pem, pkey.pem, cert.pem and certPassword.txt • Wrong certificate file type. Only files of the following types can be loaded to the system: rootCA.pem, pkey.pem and cert.pem and certPassword.txt • The contents of the certificate file does not match the system parameters
SIP TLS: Registration transport error	This alarm indicates that the communication with the SIP server cannot be established. Possible causes are: • Incorrect IP address of the SIP server • The SIP server listening port is other than the one defined in the system • The OCS services are stopped
SIP TLS: Registration handshake failure	This alarm indicates a mismatch between the security protocols of the OCS and the RMX, preventing the Registration of the RMX to the OCS.

Table B-1 *Alarms (Continued)*

Alarm Code	Alarm Description
SIP TLS: Registration server not responding	This alarm is displayed when the RMX does not receive a response from the OCS to the registration request in the expected time frame. Possible causes are: • The RMX FQDN name is not defined in the OCS pool, or is defined incorrectly. • The time frame for the expected response was too short and it will be updated with the next data refresh. The alarm may be cleared automatically the next time the data is refreshed. • The RMX FQDN name is not defined in the DNS server. Ping the DNS using the RMX FQDN name to ensure that the RMX is correctly registered to the DNS.
SIP TLS: Certificate has expired	The current TLS certificate files have expired and must be replaced with new files.
SIP TLS: Certificate is about to expire	The current TLS certificate files will expire shortly and will have to be replaced to ensure the communication with the OCS.
SIP TLS: Certificate subject name is not valid or DNS failed to resolve this name	This alarm is displayed if the name of the RMX in the certificate file is different from the FQDN name defined in the OCS.
Smart Report found errors on hard disk	Smart Report found errors on hard disk.
SSH is enabled	
Startup process failure	Process failed: [Process name] failed to start.
SWITCH not responding	
System Configuration modified	System configuration flags were modified. Reset the MCU.
Task terminated	Task terminated: [Task Name].

Table B-1 Alarms (Continued)

Alarm Code	Alarm Description
Temperature Level Critical	Possible explanations: • Temperature has reached a critical level. MCU will shut down. • Temperature problem - Critical.
Temperature Level Major	Possible explanations: • Temperature has reached a problematic level and requires attention. • Temperature problem - Major
Terminal initiated MCU reset	MCU reset was initiated by Terminal command [reset].
The Log file system is disabled	Log file system error: The Log File System is disabled. Log files not found.
The software contains patch(es)	The software contains patch(es).
Unit not responding	
Unspecified problem	Possible explanations: • Unspecified card error. • Unspecified shelf error. • Unspecified problem.
User initiated MCU reset	MCU reset was initiated by a system user.
Version upgrade is in progress	
Voltage problem	Possible reasons for the problem: • Card voltage problem. • Shelf voltage problem. • Voltage problem
Yellow Alarm	

Appendix C

CDR Fields - Unformatted File

The CDR (Call Detail Records) utility is used to retrieve conference information to a file. The CDR utility can retrieve conference information to a file in both formatted and unformatted formats.

Unformatted CDR files contain multiple records. The first record in each file contains information about the conference in general, such as the conference name and start time. The remaining records each contain information about one event that occurred during the conference, such as a participant connecting to the conference, or a user extending the length of the conference. The first field in each record identifies the event type, and this is followed by values containing information about the event. The fields are separated by commas.

Formatted files contain basically the same information as unformatted files, but with the field values replaced by descriptions. Formatted files are divided into sections, each containing information about one conference event. The first line in each section is a title describing the type of event, and this is followed by multiple lines, each containing information about the event in the form of a descriptive field name and value.

The field names and values in the formatted file will appear in the language being used for the RMX Web Client user interface at the time when the CDR information is retrieved.
The value of the fields that support Unicode values, such as the info fields, will be stored in the CDR file in UTF8. The application that reads the CDR file must support Unicode.

The MCU sends the entire CDR file via API or HTTP, and the RMX 2000 or external application does the processing and sorting. The RMX 2000 ignores events that it does not recognize, that is, events written in a higher version that do not exist in the current version. Therefore, to enable compatibility between versions, instead of adding new fields to existing events, new fields are added as separate events, so as not to affect the events from older versions. This allows users with lower versions to retrieve CDR files that were created in higher versions.

This appendix describes the fields and values in the unformatted CDR records. Although the formatted files contain basically the same information, in a few instances a single field in the unformatted file is converted to multiple lines in the formatted file, and in other cases, multiple fields in the unformatted file are combined into one line in the formatted file.

In addition, to enable compatibility for applications that were written for the MGC family, the unformatted file contains fields that were supported by the MGC family, but are not supported by the RMX 2000, whereas these fields are omitted from the formatted file.

The Conference Summary Record

The conference summary record (the first record in the unformatted CDR file) contains the following fields:

Table C-1 *Conference Summary Record Fields*

Field	Description
File Version	The version of the CDR utility that created the file.
Conference Routing Name	The Routing Name of the conference.
Internal Conference ID	The conference identification number as assigned by the system.
Reserved Start Time	Not supported. Contains the same value as the Actual Start Time field.
Reserved Duration	The amount of time the conference was scheduled to last.
Actual Start Time	The actual time the conference started in local time.
Actual Duration	The actual conference duration.
Status	The conference status code as follows: **1** - The conference is an ongoing conference. **2** - The conference was terminated by a user. **3** - The conference ended at the scheduled end time. **4** - The conference ended automatically because no participants joined the conference for a predefined time period, or all the participants disconnected from the conference and the conference was empty for a predefined time period. **5** - The conference never started. **6** - The conference could not start due to a problem. **8** - An unknown error occurred. **9** - The conference was terminated by a participant using DTMF codes. **Note:** If the conference was terminated by an MCU reset, this field will contain the value **1** (ongoing conference).

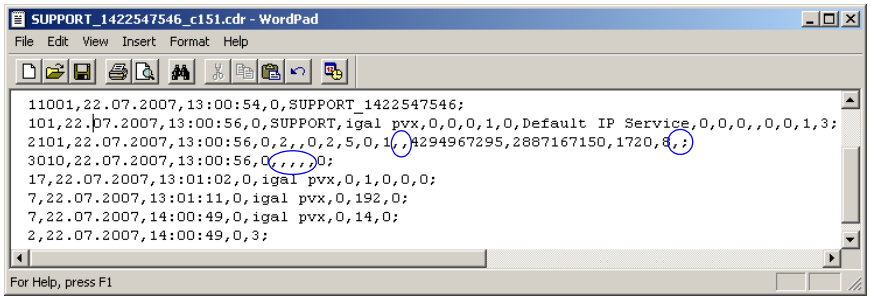
Table C-1 *Conference Summary Record Fields (Continued)*

Field	Description
File Name	The name of the conference log file.
GMT Offset Sign	Not supported. Always contains the value **0**.
GMT Offset	Not supported. Always contains the value **0**.
File Retrieved	Indicates if the file has been retrieved and saved to a formatted file, as follows: **0** - No **1** - Yes

Event Records

The event records, that is, all records in the unformatted file except the first record, contain standard fields, such as the event type code and the time stamp, followed by fields that are event specific.

The event fields are separated by commas. Two consecutive commas with nothing between them (,,), or a comma followed immediately by a semi-colon (,;), indicates an empty field, as in the example below:



Standard Event Record Fields

All event records start with the following fields:

- The CDR event type code. For a list of event type codes and descriptions, refer to Table C-2, "*CDR Event Types*", on page C-6.
- The event date.
- The event time.
- The structure length. This field is required for compatibility purposes, and always contains the value **0**.

Event Types

The table below contains a list of the events that can be logged in the CDR file, and indicates where to find details of the fields that are specific to that type of event.

The event code identifies the event in the unformatted CDR file, and the event name identifies the event in the formatted CDR file.

Table C-2 *CDR Event Types*

Event Code	Event Name	Description
1	*CONFERENCE START*	The conference started. For more information about the fields, see Table C-3, "*Event Fields for Event 1 - CONFERENCE START",* on page **C-16**. **Note:** There is one CONFERENCE START event per conference. It is always the first event in the file, after the conference summary record. It contains conference details, but not participant details.
2	*CONFERENCE END*	The conference ended. For more information about the fields, see Table C-8, "*Event Fields for Event 2 - CONFERENCE END",* on page **C-23**. **Note:** There is one CONFERENCE END event per conference, and it is always the last event in the file.
3	*ISDN/PSTN CHANNEL CONNECTED*	An ISDN/PSTN channel connected. For more information about the fields, see Table C-9, "*Event fields for Event 3 - ISDN/PSTN CHANNEL CONNECTED",* on page **C-23**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
4	*ISDN/PSTN CHANNEL DISCONNECTED*	An ISDN/PSTN channel disconnected. For more information about the fields, see Table C-10, "*Event fields for Event 4 - ISDN/PSTN CHANNEL DISCONNECTED",* on page **C-26**.
5	*ISDN/PSTN PARTICIPANT CONNECTED*	An ISDN/PSTN participant connected to the conference. For more information about the fields, see Table C-11, "*Event fields for Event 5 - ISDN/PSTN PARTICIPANT CONNECTED",* on page **C-26**.
7	*PARTICIPANT DISCONNECTED*	A participant disconnected from the conference. For more information about the fields, see Table C-12, "*Event Fields for Event 7 - PARTICIPANT DISCONNECTED",* on page **C-28**.
10	*DEFINED PARTICIPANT*	Information about a defined participant, that is, a participant who was added to the conference before the conference started. For more information about the fields, see Table C-14, "*Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT",* on page **C-30**. **Note:** There is one event for each participant defined before the conference started.
15	*H323 CALL SETUP*	Information about the IP address of the participant. For more information about the fields, see Table C-17, "*Event fields for Event 15 - H323 CALL SETUP",* on page **C-36**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
17	*H323 PARTICIPANT CONNECTED*	An H.323 participant connected to the conference. For more information about the fields, see Table C-18, "*Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED",* on page **C-37**.
18	*NEW UNDEFINED PARTICIPANT*	A new undefined participant joined the conference. For more information about the fields, see Table C-19, "*Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT",* on page **C-39**.
20	*BILLING CODE*	A billing code was entered by a participant using DTMF codes. For more information about the fields, see Table C-21, "*Event Fields for Event 20 - BILLING CODE",* on page **C-43**.
21	*SET PARTICIPANT DISPLAY NAME*	A user assigned a new name to a participant, or an end point sent its name. For more information about the fields, see Table C-22, "*Event Fields for Event 21 - SET PARTICIPANT DISPLAY NAME",* on page **C-43**.
22	*DTMF CODE FAILURE*	An error occurred when a participant entered a DTMF code. For more information about the fields, see Table C-23, "*Event Fields for Event 22 - DTMF CODE FAILURE",* on page **C-44**.

Table C-2 CDR Event Types (Continued)

Event Code	Event Name	Description
23	*SIP PARTICIPANT CONNECTED*	A SIP participant connected to the conference. For more information about the fields, see Table C-18, "*Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED",* on page **C-37**.
26	*RECORDING LINK*	A recording event, such as recording started or recording resumed, occurred. For more information about the fields, see Table C-24, "*Event fields for Event 26 - RECORDING LINK",* on page **C-44**.
28	*SIP PRIVATE EXTENSIONS*	Contains SIP Private Extensions information. For more information about the fields, see Table C-25, "*Event Fields for Event 28 - SIP PRIVATE EXTENSIONS",* on page **C-45**.
30	*GATEKEEPER INFORMATION*	Contains the gatekeeper caller ID, which makes it possible to match the CDR in the gatekeeper and in the MCU. For more information about the fields, see Table C-26, "*Event Fields for Event 30 - GATEKEEPER INFORMATION",* on page **C-46**.
31	*PARTICIPANT CONNECTION RATE*	Information about the line rate of the participant connection. This event is added to the CDR file each time the endpoint changes its connection bit rate. For more information about the fields, see Table C-27, "*Event fields for Event 31 - PARTICIPANT CONNECTION RATE",* on page **C-46**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
100	*USER TERMINATE CONFERENCE*	A user terminated the conference. For more information about the fields, see Table C-28, "*Event Fields for Event 100 - USER TERMINATE CONFERENCE",* on page **C-46**.
101	*USER ADD PARTICIPANT*	A user added a participant to the conference during the conference. For more information about the fields, see Table C-14, "*Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT",* on page **C-30**.
102	*USER DELETE PARTICIPANT*	A user deleted a participant from the conference. For more information about the fields, see Table C-29, "*Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT",* on page **C-47**.
103	*USER DISCONNECT PARTICIPANT*	A user disconnected a participant. For more information about the fields, see Table C-29, "*Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT",* on page **C-47**.
104	*USER RECONNECT PARTICIPANT*	A user reconnected a participant who was disconnected from the conference. For more information about the fields, see Table C-29, "*Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT",* on page **C-47**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
105	*USER UPDATE PARTICIPANT*	A user updated the properties of a participant during the conference. For more information about the fields, see Table C-14, "*Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT",* on page **C-30**.
106	*USER SET END TIME*	A user modified the conference end time. For more information about the fields, see Table C-30, "*Event Fields for Event 106 - USER SET END TIME",* on page **C-47**.
107	*OPERATOR MOVE PARTY FROM CONFERENCE*	The participant moved from an Entry Queue to the destination conference or between conferences. For more information about the fields, see Table C-31, "*Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY",* on page **C-47**.
108	*OPERATOR MOVE PARTY TO CONFERENCE*	The RMX User moved the participant from an ongoing conference to another conference. For more information, see Table C-32, "*Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE",* on page **C-48**.
109	*OPERATOR ATTEND PARTY*	The RMX User moved the participant to the Operator conference. For more information, see Table C-31, "*Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY",* on page **C-47**.

Table C-2 CDR Event Types (Continued)

Event Code	Event Name	Description
111	*OPERATOR BACK TO CONFERENCE PARTY*	The RMX User moved the participant back to his Home (source) conference. For more information, see Table C-33, "*Event Fields for Event 111 - OPERATOR BACK TO CONFERENCE PARTY",* on page **C-54**.
112	*OPERATOR ATTEND PARTY TO CONFERENCE*	The RMX User moved the participant from the Operator conference to another conference. For more information, see Table C-32, "*Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE",* on page **C-48**.
1001	*NEW UNDEFINED PARTICIPANT CONTINUE 1*	Additional information about a NEW UNDEFINED PARTICIPANT event. For more information about the fields, see Table C-20, "*Event Fields for Event 1001 - NEW UNDEFINED PARTY CONTINUE 1",* on page **C-43**.
2001	*CONFERENCE START CONTINUE 1*	Additional information about a CONFERENCE START event. For more information about the fields, see Table C-4, "*Event Fields for Event 2001 - CONFERENCE START CONTINUE 1",* on page **C-18**.
2007	*PARTICIPANT DISCONNECTED CONTINUE 1*	Additional information about a PARTICIPANT DISCONNECTED event. For more information about the fields, see Table C-13, "*Event Fields for Event 2007 - PARTICIPANT DISCONNECTED CONTINUE 1",* on page **C-29**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
2010	*DEFINED PARTICIPANT CONTINUE 1*	Additional information about a DEFINED PARTICIPANT event. For more information about the fields, see Table C-15, "*Event Fields for Events 2010, 2101, 2105 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1",* on page **C-33**.
2011	*DEFINED PARTICIPANT CONTINUE 2*	Additional information about a DEFINED PARTICIPANT event. For more information about the fields, see Table C-16, "*Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2102 - USER ADD PARTICIPANT CONTINUE 2, Event 2106 - USER UPDATE PARTICIPANT CONTINUE 2",* on page **C-35**.
2101	*USER ADD PARTICIPANT CONTINUE 1*	Additional information about a USER ADD PARTICIPANT event. For more information about the fields, see Table C-15, "*Event Fields for Events 2010, 2101, 2105 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1",* on page **C-33**.
2102	*USER ADD PARTICIPANT CONTINUE 2*	Additional information about a USER ADD PARTICIPANT event. For more information about the fields, see Table C-16, "*Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2102 - USER ADD PARTICIPANT CONTINUE 2, Event 2106 - USER UPDATE PARTICIPANT CONTINUE 2",* on page **C-35**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
2105	*USER UPDATE PARTICIPANT CONTINUE 1*	Additional information about a USER UPDATE PARTICIPANT event. For more information about the fields, see Table C-15, "*Event Fields for Events 2010, 2101, 2105 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1",* on page **C-33**.
2106	*USER UPDATE PARTICIPANT CONTINUE 2*	Additional information about a USER UPDATE PARTICIPANT event. For more information about the fields, see Table C-16, "*Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2102 - USER ADD PARTICIPANT CONTINUE 2, Event 2106 - USER UPDATE PARTICIPANT CONTINUE 2",* on page **C-35**.
3010	*PARTICIPANT INFORMATION*	The contents of the participant information fields. For more information about the fields, see Table C-34, "*Event Fields for Event 3010 - PARTICIPANT INFORMATION",* on page **C-54**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
5001	*CONFERENCE START CONTINUE 4*	Additional information about a CONFERENCE START event. For more information about the fields, see Table C-5, "*Event Fields for Event 5001 - CONFERENCE START CONTINUE 4",* on page **C-21**. **Note:** An additional CONFERENCE START CONTINUE 4 event will be written to the CDR each time the value of one of the following conference fields is modified: • Conference Password • Chairperson Password • Info1, Info2 or Info3 • Billing Info These additional events will only contain the value of the modified field.
6001	*CONFERENCE START CONTINUE 5*	Additional information about a CONFERENCE START event. For more information about the fields, see Table C-6, "*Event Fields for Event 6001 - CONFERENCE START CONTINUE 5",* on page **C-22**.
11001	*CONFERENCE START CONTINUE 10*	Additional information about a CONFERENCE START event. This event contains the Display Name. For more information about the fields, see Table C-7, "*Event Fields for Event 11001 - CONFERENCE START CONTINUE 10",* on page **C-22**.

This list only includes events that are supported by the RMX 2000. For a list of MGC Manager events that are not supported by the RMX 2000, see "*MGC Manager Events that are not Supported by the RMX 2000"* on page **C-59**.

Event Specific Fields

The following tables describe the fields which are specific to each type of event.

Some fields that were supported by the MGC Manager, are not supported by the RMX 2000. In addition, for some fields the RMX 2000 has a fixed value, whereas the MGC Manager supported multiple values. For more information about the MGC Manager fields and values, see the *MGC Manager User's Guide Volume II*, *Appendix A*.

Table C-3 *Event Fields for Event 1 - CONFERENCE START*

Field	Description
Dial-Out Manually	Indicates whether the conference was a dial-out manually conference or not. Currently the only value is: **0** - The conference was *not* a dial-out manually conference, that is, the MCU initiates the communication with dial-out participants, and the user does not need to connect them manually.
Auto Terminate	Indicates whether the conference was set to end automatically if no participant joins the conference for a predefined time period after the conference starts, or if all participants disconnect from the conference and the conference is empty for a predefined time period. Possible values are: **0** - The conference was *not* set to end automatically. **1** - The conference was set to end automatically.
Line Rate	The conference line rate, as follows: **0** - 64 kbps **6** - 384 kbps **12** - 1920 kbps **13** - 128 kbps **15** - 256 kbps **23** - 512 kbps **24** - 768 kbps **26** - 1152 kbps **29** - 1472 kbps **32** - 96 kbps

Table C-3 *Event Fields for Event 1 - CONFERENCE START (Continued)*

Field	Description
Line Rate (cont.)	**33** - 1024 kbps **34** - 4096 kbps
Restrict Mode	Not supported. Always contains the value **0**.
Audio Algorithm	The audio algorithm. Currently the only value is: **255** - Auto
Video Session	The video session type. Currently the only value is: **3** - Continuous Presence
Video Format	The video format. Currently the only value is: **255** - Auto
CIF Frame Rate	The CIF frame rate. Currently the only value is: **255** -Auto
QCIF Frame Rate	The QCIF frame rate: Currently the only value is: **255** - Auto
LSD Rate	Not supported. Always contains the value **0**.
HSD Rate	Not supported. Always contains the value **0**.
T120 Rate	Not supported. Always contains the value **0**.

Table C-4 Event Fields for Event 2001 - CONFERENCE START CONTINUE 1

Field	Description
Audio Tones	Not supported. Always contains the value **0**.
Alert Tone	Not supported. Always contains the value **0**.
Talk Hold Time	The minimum time that a speaker has to speak to become the video source. The value is in units of 0.01 seconds. Currently the only value is **150**, which indicates a talk hold time of 1.5 seconds.
Audio Mix Depth	The maximum number of participants whose audio can be mixed. Currently the only value is **5**.
Operator Conference	Not supported. Always contains the value **0**.
Video Protocol	The video protocol. Currently the only value is: **255** - Auto
Meet Me Per Conference	Indicates the Meet Me Per Conference setting. Currently the only value is: **1** - The Meet Me Per Conference option is enabled, and dial-in participants can join the conference by dialing the dial-in number.
Number of Network Services	Not supported. Always contains the value **0**.
Chairperson Password	The chairperson password for the conference. An empty field “” means that no chairperson password was assigned to the conference.
Chair Mode	Not supported. Always contains the value **0**.

Table C-4 *Event Fields for Event 2001 - CONFERENCE START CONTINUE 1 (Continued)*

Field	Description
Cascade Mode	The cascading mode. Currently the only value is: **0** - None
Master Name	Not supported. This field remains empty.
Minimum Number of Participants	The number of participants for which the system reserved resources. Additional participants may join the conference without prior reservation until all the resources are utilized. Currently the only value is **0**.
Allow Undefined Participants	Indicates whether or not undefined dial-in participants can connect to the conference. Currently the only value is: **1** - Undefined participants can connect to the conference
Time Before First Participant Joins	**Note:** This field is only relevant if the Auto Terminate option is enabled. Indicates the number of minutes that should elapse from the time the conference starts, without any participant connecting to the conference, before the conference is automatically terminated by the MCU.
Time After Last Participant Quits	**Note:** This field is only relevant if the Auto Terminate option is enabled. Indicates the number of minutes that should elapse after the last participant has disconnected from the conference, before the conference is automatically terminated by the MCU.
Conference Lock Flag	Not supported. Always contains the value **0**.

Table C-4 Event Fields for Event 2001 - CONFERENCE START CONTINUE 1 (Continued)

Field	Description
Maximum Number of Participants	The maximum number of participants that can connect to the conference at one time. The value **65535** (auto) indicates that as many participants as the MCU's resources allow can connect to the conference, up to the maximum possible for the type of conference.
Audio Board ID	Not supported. Always contains the value **65535**.
Audio Unit ID	Not supported. Always contains the value **65535**.
Video Board ID	Not supported. Always contains the value **65535**.
Video Unit ID	Not supported. Always contains the value **65535**.
Data Board ID	Not supported. Always contains the value **65535**.
Data Unit ID	Not supported. Always contains the value **65535**.
Message Service Type	The Message Service type. Currently the only value is: **3** - IVR
Conference IVR Service	The name of the IVR Service assigned to the conference. **Note:** If the name of the IVR Service contains more than 20 characters, it will be truncated to 20 characters.
Lecture Mode Type	Indicates the type of Lecture Mode, as follows: **0** - None **1** - Lecture Mode **3** - Presentation Mode

Table C-4 *Event Fields for Event 2001 - CONFERENCE START CONTINUE 1 (Continued)*

Field	Description
Lecturer	**Note:** This field is only relevant if the Lecture Mode Type is Lecture Mode. The name of the participant selected as the conference lecturer.
Time Interval	**Note:** This field is only relevant if Lecturer View Switching is enabled. The number of seconds a participant is to be displayed in the lecturer window before switching to the next participant. Currently the only value is **15**.
Lecturer View Switching	**Note:** This field is only relevant when Lecture Mode is enabled. Indicates the lecturer view switching setting, as follows: **0** - Automatic switching between participants is disabled. **1** - Automatic switching between participants is enabled.
Audio Activated	Not supported. Always contains the value **0**.
Lecturer ID	Not supported. Always contains the value **4294967295**.

Table C-5 *Event Fields for Event 5001 - CONFERENCE START CONTINUE 4*

Field	Description
Note: When this event occurs as the result of a change to the value of one of the event fields, the event will only contain the value of the modified field. All other fields will be empty.	
Conference ID	The conference ID.

Table C-5 *Event Fields for Event 5001 - CONFERENCE START CONTINUE 4 (Continued)*

Field	Description
Conference Password	The conference password. An empty field "" means that no conference password was assigned to the conference.
Chairperson Password	The chairperson password. An empty field "" means that no chairperson password was assigned to the conference.
Info1 *Info2* *Info3*	The contents of the conference information fields. These fields enable users to enter general information for the conference, such as the company name, and the contact person's name and telephone number. The maximum length of each field is 80 characters.
Billing Info	The billing code.

Table C-6 *Event Fields for Event 6001 - CONFERENCE START CONTINUE 5*

Field	Description
Encryption	Indicates the conference encryption setting, as follows: **0** - The conference is *not* encrypted. **1** - The conference is encrypted.

Table C-7 *Event Fields for Event 11001 - CONFERENCE START CONTINUE 10*

Field	Description
Display Name	The Display Name of the conference.

Table C-8 *Event Fields for Event 2 - CONFERENCE END*

Field	Description
Conference End Cause	Indicates the reason for the termination of the conference, as follows: **1** - The conference is an ongoing conference or the conference was terminated by an MCU reset. **2** - The conference was terminated by a user. **3** - The conference ended at the scheduled end time. **4** - The conference ended automatically because no participants joined the conference for a predefined time period, or all the participants disconnected from the conference and the conference was empty for a predefined time period. **5** - The conference never started. **6** - The conference could not start due to a problem. **8** - An unknown error occurred. **9** - The conference was terminated by a participant using DTMF codes.

Table C-9 *Event fields for Event 3 - ISDN/PSTN CHANNEL CONNECTED*

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Channel ID	The channel identifier.
Number of Channels	The number of channels being connected for this participant.
Connect Initiator	Indicates who initiated the connection, as follows: **0** - RMX **1** - Participant **Any other number** - Unknown

Table C-9 Event fields for Event 3 - ISDN/PSTN CHANNEL CONNECTED (Continued)

Field	Description
Call Type	The call type, as follows: **68** - 56 KBS data call **72** - 1536kbs data call (PRI only) **75** - 56 KBS data call **77** - Modem data service **79** - 384kbs data call (PRI only) **86** - Normal voice call
Network Service Program	The Network Service program, as follows: **0** - None **1** - ATT_SDN or NTI_PRIVATE **3** - ATT_MEGACOM or NTI_OUTWATS **4** - NTI FX **5** - NTI TIE TRUNK **6** - ATT ACCUNET **8** - ATT 1800 **16** - NTI_TRO
Preferred Mode	The value of the preferred/exclusive field for B channel selection (the PRF mode), as follows: **0** - None **1** - Preferred **2** - Exclusive For more details refer to the Q.931 standard.
Calling Participant Number Type	The type of calling number, as follows: **0** - Unknown, default **1** - International **2** - National **3** - Network specific **4** - Subscriber **6** - Abbreviated
Calling Participant Number Plan	The calling participant number plan. **0** - Unknown **1** - ISDN/PSTN **9** - Private

Table C-9 *Event fields for Event 3 - ISDN/PSTN CHANNEL CONNECTED (Continued)*

Field	Description
Calling Participant Presentation Indicator	The calling participant presentation indicator, as follows: **0** - Presentation allowed, default **1** - Presentation restricted **2** - Number not available **255** - Unknown
Calling Participant Screening Indicator	The calling participant screening indicator, as follows: **0** - Participant not screened, default **1** - Participant verification succeeded **2** - Participant verification failed **3** - Network provided **255** - Unknown
Calling Participant Phone Number	The telephone number used for dial-in.
Called Participant Number Type	The type of number called, as follows: **0** - Unknown, default **1** - International **2** - National **3** - Network specific **4** - Subscriber **6** - Abbreviated
Called Participant Number Plan	The called participant number plan, as follows: **0** - Unknown **1** - ISDN/PSTN **9** - Private
Called Participant Phone Number	The telephone number used for dial-out.

Table C-10 Event fields for Event 4 - ISDN/PSTN CHANNEL DISCONNECTED

Field	Description
Participant Name	The participant name.
Participant ID	The identification number assigned to the participant by the MCU.
Channel ID	The channel identifier.
Disconnect Initiator	Indicates who initiated the disconnection, as follows: **0** - RMX **1** - Participant **Any other number** - Unknown
Disconnect Coding Standard	The disconnection cause code standard. For values and explanations, see the Q.931 Standard.
Disconnect Location	The disconnection cause location. For values and explanations, see the Q.931 Standard.
Q931 Disconnection Cause	The disconnection cause value. For values and explanations, see the Q.931 Standard.

Table C-11 Event fields for Event 5 - ISDN/PSTN PARTICIPANT CONNECTED

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.

Table C-11 *Event fields for Event 5 - ISDN/PSTN PARTICIPANT CONNECTED (Continued)*

Field	Description
Participant Status	The participant status, as follows: **0** - Idle **1** - Connected **2** - Disconnected **3** - Waiting for dial-in **4** - Connecting **5** - Disconnecting **6** - Partially connected. Party has completed H.221 capability exchange **7** - Deleted by a user **8** - Secondary. The participant could not connect the video channels and is connected via audio only **10** - Connected with problem **11** - Redialing
Remote Capabilities	**Note:** This field is only relevant to ISDN video participants. The remote capabilities in H.221 format.
Remote Communication Mode	**Note:** This field is only relevant to ISDN video participants. The remote communication mode in H.221 format.
Secondary Cause	**Note:** This field is only relevant to ISDN video participants and only if the Participant Status is Secondary. The cause for the secondary connection (not being able to connect the video channels), as follows: **0** - Default **11** - The incoming video parameters are not compatible with the conference video parameters **12** - H.323 card failure **13** - The conference video settings are not compatible with the endpoint capabilities **14** - The new conference settings are not compatible with the endpoint capabilities

Table C-11 Event fields for Event 5 - ISDN/PSTN PARTICIPANT CONNECTED (Continued)

Field	Description
Secondary Cause (cont.)	**15** - Video stream violation due to incompatible annexes or other discrepancy. **16** - Inadequate video resources **17** - When moved to a Transcoding or Video Switching conference, the participant's video capabilities are not supported by the video cards **18** - Video connection could not be established **24** - The endpoint closed its video channels **25** - The participant video settings are not compatible with the conference protocol **26** - The endpoint could not re-open the video channel after the conference video mode was changed **27** - The gatekeeper approved a lower bandwidth than requested **28** - Video connection for the SIP participant is temporarily unavailable **29** - AVF problem. Insufficient bandwidth. **30** - H2.39 bandwidth mismatch **255** - Other

Table C-12 Event Fields for Event 7 - PARTICIPANT DISCONNECTED

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Call Disconnection Cause	The disconnection cause. For more information about possible values, see Table C-35, "*Disconnection Cause Values",* on page **C-55**.
Q931 Disconnect Cause	If the disconnection cause is "No Network Connection" or "Participant Hang Up", then this field indicates the Q931 disconnect cause.

Table C-13 Event Fields for Event 2007 - PARTICIPANT DISCONNECTED CONTINUE 1

Field	Description
Rx Synchronization Loss	The number of times that the general synchronization of the MCU was lost.
Tx Synchronization Loss	The number of times that the general synchronization of the participant was lost.
Rx Video Synchronization Loss	The number of times that the synchronization of the MCU video unit was lost.
Tx Video Synchronization Loss	The number of times that the synchronization of the participant video was lost.
Mux Board ID	Not supported. Always contains the value **0**.
Mux Unit ID	Not supported. Always contains the value **0**.
Audio Codec Board ID	Not supported. Always contains the value **0**.
Audio Codec Unit ID	Not supported. Always contains the value **0**.
Audio Bridge Board ID	Not supported. Always contains the value **0**.
Audio Bridge Unit ID	Not supported. Always contains the value **0**.
Video Board ID	Not supported. Always contains the value **0**.
Video Unit ID	Not supported. Always contains the value **0**.
T.120 Board ID	Not supported. Always contains the value **0**.

Table C-13 *Event Fields for Event 2007 - PARTICIPANT DISCONNECTED CONTINUE 1*

Field	Description
T.120 Unit ID	Not supported. Always contains the value **0**.
T.120 MCS Board ID	Not supported. Always contains the value **0**.
T.120 MCS Unit ID	Not supported. Always contains the value **0**.
H.323 Board ID	Not supported. Always contains the value **0**.
H323 Unit ID	Not supported. Always contains the value **0**.

Table C-14 *Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT*

Field	Description
User Name	The login name of the user who added the participant to the conference, or updated the participant properties.
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Dialing Direction	The dialing direction, as follows: **0** - Dial-out **5** - Dial-in
Bonding Mode	Not supported. Always contains the value **0**.

Table C-14 *Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT (Continued)*

Field	Description
Number Of Channels	**Note:** This field is only relevant to ISDN/PSTN participants. The number of channels being connected for this participant.
Net Channel Width	Not supported. Always contains the value **0**.
Network Service Name	The name of the Network Service. An empty field "" indicates the default Network Service.
Restrict	Not supported. Always contains the value **0**.
Audio Only	Indicates the participant's Audio Only setting, as follows: **0** - The participant is *not* an Audio Only participant **1** - The participant is an Audio Only participant **255** - Unknown
Default Number Type	**Note:** This field is only relevant to ISDN/PSTN participants. The type of telephone number, as follows: **0** - Unknown **1** - International **2** - National **3** - Network specific **4** - Subscriber **6** - Abbreviated **255** - Taken from Network Service, default **Note:** For dial-in participants, the only possible value is: **255** - Taken from Network Service
Net Sub-Service Name	Not supported. This field remains empty.

Table C-14 Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT (Continued)

Field	Description
Number of Participant Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The number of participant phone numbers. In a dial-in connection, the participant phone number is the CLI (Calling Line Identification) as identified by the MCU. In a dial-out connection, participant phone numbers are the phone numbers dialed by the MCU for each participant channel.
Number of MCU Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The number of MCU phone numbers. In a dial-in connection, the MCU phone number is the number dialed by the participant to connect to the MCU. In a dial-out connection, the MCU phone number is the MCU (CLI) number as seen by the participant.
Party and MCU Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. No, one or more fields, one field for each participant and MCU phone number. The participant phone numbers are listed first, followed by the MCU phone numbers.
Identification Method	**Note:** This field is only relevant to dial-in participants. The method by which the destination conference is identified, as follows: **1** - Called phone number, IP address or alias **2** - Calling phone number, IP address or alias
Meet Me Method	**Note:** This field is only relevant to dial-in participants. The meet-me per method. Currently the only value is: **3** - Meet-me per participant

Table C-15 Event Fields for Events 2010, 2101, 2105 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1

Field	Description
Network Type	The type of network between the participant and the MCU, as follows: **0** - ISDN/PSTN **2** - H.323 **5** - SIP
H.243 Password	Not supported. This field remains empty.
Chair	Not supported. Always contains the value **0**.
Video Protocol	The video protocol used by the participant, as follows: **1** - H.261 **2** - H.263 **4** - H.264 **255** - Auto
Broadcasting Volume	The broadcasting volume assigned to the participant. The value is between **1** (lowest) and **10** (loudest). Each unit movement increases or decreases the volume by **3 dB**.
Undefined Participant	Indicates whether are not the participant is an undefined participant, as follows: **0** - The participant is *not* an undefined participant. **2** - The participant is an undefined participant.
Node Type	The node type, as follows: **0** - MCU **1** - Terminal

Table C-15 Event Fields for Events 2010, 2101, 2105 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1 (Continued)

Field	Description
Bonding Phone Number	**Note:** This field is only relevant to ISDN/PSTN participants. The phone number for Bonding dial-out calls. Bonding is a communication protocol that aggregates from two up to thirty 64 Kbps B channels together, to look like one large bandwidth channel.
Video Bit Rate	The video bit rate in units of kilobits per second. A value of **4294967295** denotes auto, and in this case, the rate is computed by the MCU.
IP Address	**Note:** This field is only relevant to IP participants. The IP address of the participant. An address of **4294967295** indicates that no IP address was specified for the participant, and the gatekeeper is used for routing. In all other cases the address overrides the gatekeeper.
Signaling Port	**Note:** This field is only relevant to IP participants. The signaling port used for participant connection.
H.323 Participant Alias Type/SIP Participant Address Type	**Note:** This field is only relevant to IP participants. For H.323 participants, the alias type, as follows: **7** - E164 **8** - H.323 ID **13** - Email ID **14** - Participant number For SIP participants, the address type, as follows: **1** - SIP URI **2** - Tel URL

Table C-15 *Event Fields for Events 2010, 2101, 2105 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1 (Continued)*

Field	Description
H.323 Participant Alias Name/SIP Participant Address	**Note:** This field is only relevant to IP participants. For H.323 participants: The participant alias. The alias may contain up to 512 characters. For SIP participants: The participant address. The address may contain up to 80 characters.

Table C-16 *Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2102 - USER ADD PARTICIPANT CONTINUE 2, Event 2106 - USER UPDATE PARTICIPANT CONTINUE 2*

Field	Description
Encryption	Indicates the participant's encryption setting as follows: **0** - The participant is *not* encrypted. **1** - The participant is encrypted. **2** - Auto. The conference encryption setting is applied to the participant.
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.

Table C-17 Event fields for Event 15 - H323 CALL SETUP

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Connect Initiator	Indicates who initiated the connection, as follows: **0** - MCU **1** - Remote participant **Any other number** - Unknown
Min Rate	The minimum line rate used by the participant. The data in this field should be ignored. For accurate rate information, see CDR event 31.
Max Rate	The maximum line rate achieved by the participant. The data in this field should be ignored. For accurate rate information, see CDR event 31.
Source Party Address	The IP address of the calling participant. A string of up to 255 characters.
Destination Party Address	The IP address of the called participant. A string of up to 255 characters.
Endpoint Type	The endpoint type, as follows: **0** - Terminal **1** - Gateway **2** - MCU **3** - Gatekeeper **4** - Undefined

Table C-18 *Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED*

Field	Description
Participant Name	The name of the participant. An empty field "" denotes an unidentified participant or a participant whose name is unspecified.
Participant ID	The identification number assigned to the participant by the MCU.
Participant Status	The participant status, as follows: **0** - Idle **1** - Connected **2** - Disconnected **3** - Waiting for dial-in **4** – Connecting **5** - Disconnecting **6** - Partially connected. Party has completed H.221 capability exchange **7** - Deleted by a user **8** - Secondary. The participant could not connect the video channels and is connected via audio only **10** - Connected with problem **11** - Redialing
Capabilities	Not supported. Always contains the value **0**.
Remote Communication Mode	Not supported. Always contains the value **0**.

Table C-18 Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED (Continued)

Field	Description
Secondary Cause	**Note:** This field is only relevant if the Participant Status is Secondary. The cause for the secondary connection (not being able to connect the video channels), as follows: **0** - Default. **11** - The incoming video parameters are not compatible with the conference video parameters **13** - The conference video settings are not compatible with the endpoint capabilities **14** - The new conference settings are not compatible with the endpoint capabilities **15** - Video stream violation due to incompatible annexes or other discrepancy **16** - Inadequate video resources **17** - When moved to a Transcoding or Video Switching conference, the participant's video capabilities are not supported by the video cards **18** - Video connection could not be established **24** - The endpoint closed its video channels **25** - The participant video settings are not compatible with the conference protocol **26** - The endpoint could not re-open the video channel after the conference video mode was changed **27** - The gatekeeper approved a lower bandwidth than requested **28** - Video connection for the SIP participant is temporarily unavailable **255** - Other

Table C-19 *Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT*

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Dialing Direction	The dialing direction, as follows **0** - Dial-out **5** - Dial-in
Bonding Mode	Not supported. Always contains the value **0**.
Number of Channels	**Note:** This field is only relevant to ISDN/PSTN participants. The number of channels being connected for this participant.
Net Channel Width	Not supported. Always contains the value **0**.
Network Service Name	The name of the Network Service. An empty field "" indicates the default Network Service.
Restrict	Not supported. Always contains the value **0**.
Audio Only	Indicates the participant's Audio Only setting, as follows: **0** - The participant is *not* an Audio Only participant **1** - The participant is an Audio Only participant **255** - Unknown
Default Number Type	**Note:** This field is only relevant to ISDN/PSTN participants. The type of telephone number. **Note:** Since undefined participants are always dial-in participants, the only possible value is: **255** - Taken from Network Service

Table C-19 Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT (Continued)

Field	Description
Net Sub-Service Name	Not supported. This field remains empty.
Number of Participant Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The number of participant phone numbers. The participant phone number is the CLI (Calling Line Identification) as identified by the MCU.
Number of MCU Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The number of MCU phone numbers. The MCU phone number is the number dialed by the participant to connect to the MCU.
Party and MCU Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. No, one or more fields, one field for each participant and MCU phone number. The participant phone numbers are listed first, followed by the MCU phone numbers.
Identification Method	**Note:** This field is only relevant to dial-in participants. The method by which the destination conference is identified, as follows: **1** - Called phone number, IP address or alias **2** - Calling phone number, IP address or alias
Meet Me Method	**Note:** This field is only relevant to dial-in participants. The meet-me per method, as follows: **3** - Meet-me per participant
Network Type	The type of network between the participant and the MCU, as follows: **0** - ISDN/PSTN **2** - H.323 **5** - SIP

Table C-19 *Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT (Continued)*

Field	Description
H.243 Password	Not supported. This field remains empty.
Chair	Not supported. Always contains the value **0**.
Video Protocol	The video protocol, as follows: **1** - H.261 **2** - H.263 **4** - H.264 **255** - Auto
Broadcasting Volume	The broadcasting volume assigned to the participant. The value is between **1** (lowest) and **10** (loudest). Each unit movement increases or decreases the volume by **3 dB**.
Undefined Participant	Indicates whether are not the participant is an undefined participant, as follows: **0** - The participant is *not* an undefined participant. **2** - The participant is an undefined participant.
Node Type	The node type, as follows: **0** - MCU **1** - Terminal
Bonding Phone Number	**Note:** This field is only relevant to ISDN/PSTN participants. The phone number for Bonding dial-out calls. Bonding is a communication protocol that aggregates from two up to thirty 64 Kbps B channels together, to look like one large bandwidth channel.
Video Bit Rate	The video bit rate in units of kilobits per second. A value of **4294967295** denotes auto, and in this case, the rate is computed by the MCU.

Table C-19 Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT (Continued)

Field	Description
IP Address	**Note:** This field is only relevant to IP participants. The IP address of the participant. An address of **4294967295** indicates that no IP address was specified for the participant, and the gatekeeper is used for routing. In all other cases the address overrides the gatekeeper.
Signaling Port	**Note:** This field is only relevant to IP participants. The signaling port used for participant connection. A value of **65535** is ignored by MCU.
H.323 Participant Alias Type/SIP Participant Address Type	**Note:** This field is only relevant to IP participants. For H.323 participants, the alias type, as follows: **7** - E164 **8** - H.323 ID **13** - Email ID **14** - Participant number For SIP participants, the address type, as follows: **1** - SIP URI **2** - Tel URL
H.323 Participant Alias Name/SIP Participant Address	**Note:** This field is only relevant to IP participants. For H.323 participants: The participant alias. The alias may contain up to 512 characters. For SIP participants: The participant address. The address may contain up to 80 characters.

Table C-20 *Event Fields for Event 1001 - NEW UNDEFINED PARTY CONTINUE 1*

Field	Description
Encryption	Indicates the participant's encryption setting as follows: **0** - The participant is *not* encrypted. **1** - The participant is encrypted. **2** - Auto. The conference encryption setting is applied to the participant.
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.

Table C-21 *Event Fields for Event 20 - BILLING CODE*

Field	Description
Participant Name	The name of the participant who added the billing code.
Participant ID	The identification number, as assigned by the MCU, of the participant who added the billing code.
Billing Info	The numeric billing code that was added (32 characters).

Table C-22 *Event Fields for Event 21 - SET PARTICIPANT DISPLAY NAME*

Field	Description
Participant Name	The original name of the participant, for example, the name automatically assigned to an undefined participant, such as, "<conference name>_(000)".
Participant ID	The identification number assigned to the participant by the MCU.
Display Name	The new name assigned to the participant by the user, or the name sent by the end point.

Table C-23 Event Fields for Event 22 - DTMF CODE FAILURE

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Incorrect Data	The incorrect DTMF code entered by the participant, or an empty field "" if the participant did not press any key.
Correct Data	The correct DTMF code, if known.
Failure Type	The type of DTMF failure, as follows: **2** - The participant did not enter the correct conference password. **6** - The participant did not enter the correct chairperson password. **12** - The participant did not enter the correct Conference ID.

Table C-24 Event fields for Event 26 - RECORDING LINK

Field	Description
Participant Name	The name of the Recording Link participant.
Participant ID	The identification number assigned to the Recording Link participant by the MCU.
Recording Operation	The type of recording operation, as follows: **0** - Start recording **1** - Stop recording **2** - Pause recording **3** - Resume recording **4** - Recording ended **5** - Recording failed
Initiator	Not supported.

Table C-24 Event fields for Event 26 - RECORDING LINK (Continued)

Field	Description
Recording Link Name	The name of the Recording Link.
Recording Link ID	The Recording Link ID.
Start Recording Policy	The start recording policy, as follows: **1** - Start recording automatically as soon as the first participant connects to the conference. **2** - Start recording when requested by the conference chairperson via DTMF codes or from the RMX Web Client, or when the operator starts recording from the RMX Web Client.

Table C-25 Event Fields for Event 28 - SIP PRIVATE EXTENSIONS

Field	Description
Participant Name	The name of the participant.
Participant ID	The participant's identification number as assigned by the system.
Called Participant ID	The called participant ID.
Asserted Identity	The identity of the user sending a SIP message as it was verified by authentication.
Charging Vector	A collection of charging information.
Preferred Identity	The identity the user sending the SIP message wishes to be used for the P-Asserted-Header field that the trusted element will insert.

Table C-26 Event Fields for Event 30 - GATEKEEPER INFORMATION

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Gatekeeper Caller ID	The caller ID in the gatekeeper records. This value makes it possible to match the CDR in the gatekeeper and in the MCU.

Table C-27 Event fields for Event 31 - PARTICIPANT CONNECTION RATE

Field	Description
Participant Name	The participant name.
Participant ID	The identification number assigned to the participant by the MCU.
Participant Current Rate	The participant line rate in Kbps.

Table C-28 Event Fields for Event 100 - USER TERMINATE CONFERENCE

Field	Description
Terminated By	The login name of the user who terminated the conference.

Table C-29 *Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT*

Field	Description
User Name	The login name of the user who reconnected the participant to the conference, or disconnected or deleted the participant from the conference.
Participant Name	The name of the participant reconnected to the conference, or disconnected or deleted from the conference.
Participant ID	The identification number assigned to the participant by the MCU.

Table C-30 *Event Fields for Event 106 - USER SET END TIME*

Field	Description
New End Time	The new conference end time set by the user, in GMT time.
User Name	The login name of the user who changed the conference end time.

Table C-31 *Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY*

Field	Description
Operator Name	The login name of the user who moved the participant.
Party Name	The name of the participant who was moved.
Party ID	The identification number of the participant who was moved, as assigned by the MCU.

Table C-31 Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY (Continued) (Continued)

Field	Description
Destination Conf Name	The name of the conference to which the participant was moved.
Destination Conf ID	The identification number of the conference to which the participant was moved.

Table C-32 Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE

Field	Description
Operator Name	The login name of the operator who moved the participant to the conference.
Source Conf Name	The name of the source conference.
Source Conf ID	The identification number of the source conference, as assigned by the MCU.
Party Name	The name of the participant who was moved.
Party ID	The identification number assigned to the participant by the MCU.
Connection Type	The connection type, as follows: **0** - Dial-out **5** - Dial-in
Bonding Mode	**Note:** This field is only relevant to ISDN/PSTN participants. Possible values are: **0** - Bonding is disabled **1** - Bonding is enabled **255** - Auto

Table C-32 *Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE (Continued)*

Field	Description
Number Of Channels	**Note:** This field is only relevant to ISDN/PSTN participants. The number of channels, as follows: **255** - Auto Otherwise, in range of **1** - **30**
Net Channel Width	The bandwidth of each channel. This value is always **0**, which represents a bandwidth of **1B**, which is the only bandwidth that is currently supported.
Net Service Name	The name of the Network Service. An empty field "" indicates the default Network Service.
Restrict	Indicates whether or not the line is restricted, as follows: **27** - Restricted line **28** - Non restricted line **255** - Unknown or not relevant
Voice Mode	Indicates whether or not the participant is an Audio Only participant, as follows: **0** - The participant is *not* an Audio Only participant **1** - The participant is an Audio Only participant **255** - Unknown

Table C-32 Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE (Continued)

Field	Description
Number Type	**Note:** This field is only relevant to dial-out participants. **Note:** This field is only relevant to ISDN/PSTN participants. The type of telephone number, as follows: **0** - Unknown **1** - International **2** - National **3** - Network specific **4** - Subscriber **6** - Abbreviated **255** - Taken from Network Service, default
Net SubService Name	**Note:** This field is only relevant to dial-out participants. **Note:** This field is only relevant to ISDN/PSTN participants. The network sub-service name. An empty field “” means that MCU selects the default sub-service.
Number of Party Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The number of participant phone numbers. In a dial-in connection, the participant phone number is the CLI (Calling Line Identification) as identified by the MCU. In a dial-out connection, participant phone numbers are the phone numbers dialed by the MCU for each participant channel.

Table C-32 Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE (Continued)

Field	Description
Number of MCU Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The number of MCU phone numbers. In a dial-in connection, the MCU phone number is the number dialed by the participant to connect to the MCU. In a dial-out connection, the MCU phone number is the MCU (CLI) number as seen by the participant.
Party and MCU Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The participant phone numbers are listed first, followed by the MCU phone numbers.
Ident. Method	**Note:** This field is only relevant to dial-in participants. The method by which the destination conference is identified, as follows: **0** - Password **1** - Called phone number, or IP address, or alias **2** - Calling phone number, or IP address, or alias
Meet Method	**Note:** This field is only relevant to dial-in participants. The meet-me per method, as follows: **1** - Meet-me per MCU-Conference **3** - Meet-me per participant **4** - Meet-me per channel
Net Interface Type	The type of network interface between the participant and the MCU, as follows: **0** - ISDN **2** - H.323 **5** - SIP
H243 Password	The H.243 password, or an empty field "" if there is no password.

Table C-32 Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE (Continued)

Field	Description
Chair	Not supported. Always contains the value **0**.
Video Protocol	The video protocol, as follows: **1** - H.261 **2** - H.263 **3** - H.264* **4** - H.264 **255** - Auto
Audio Volume	The broadcasting volume assigned to the participant. The value is between **1** (lowest) and **10** (loudest).
Undefined Type	The participant type, as follows: **0** - Defined participant. (The value in the formatted text file is “default”.) **2** - Undefined participant. (The value in the formatted text file is “Unreserved participant “.)
Node Type	The node type, as follows: **0** - MCU **1** - Terminal
Bonding Phone Number	**Note:** This field is only relevant to ISDN/PSTN participants. The phone number for Bonding dial-out calls.
Video Rate	**Note:** This field is only relevant to IP participants. The video rate in units of kilobits per second. A value of **4294967295** denotes auto, and in this case, the rate is computed by the MCU.

Table C-32 *Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE (Continued)*

Field	Description
IP Address	**Note:** This field is only relevant to IP participants. The IP address of the participant. An address of **4294967295** indicates that no IP address was specified for the participant, and the gatekeeper is used for routing. In all other cases the address overrides the gatekeeper.
Call Signaling Port	**Note:** This field is only relevant to IP participants. The signaling port used for participant connection. A value of **65535** is ignored by MCU.
H.323 Party Alias Type/SIP Party Address Type	**Note:** This field is only relevant to IP participants. For H.323 participants, the alias type, as follows: **7** - E164 **8** - H.323 ID **11** - URL ID alias type **12** - Transport ID **13** - Email ID **14** - Participant number For SIP participants, the address type, as follows: **1** - SIP URI **2** - Tel URL
H.323 Party Alias/ SIP Party Address	**Note:** This field is only relevant to IP participants. For H.323 participants, the participant alias. The alias may contain up to 512 characters. For SIP participants, the participant address. The address may contain up to 80 characters.

Table C-33 Event Fields for Event 111 - OPERATOR BACK TO CONFERENCE PARTY

Field	Description
Operator Name	The login name of the operator moving the participant back to the conference.
Party Name	The name of the participant being moved.
Party ID	The identification number, as assigned by the MCU, of the participant being moved.

Table C-34 Event Fields for Event 3010 - PARTICIPANT INFORMATION

Field	Description
Info1 *Info2* *Info3* *Info4*	The participant information fields. These fields enable users to enter general information about the participant, such as the participant's e-mail address. The maximum length of each field is 80 characters.
VIP	Not supported. Always contains the value **0**.

Disconnection Cause Values

For an explanation of the disconnection causes, see *Appendix A: "Disconnection Causes"* on page **A-1**.

Table C-35 *Disconnection Cause Values*

Value	Call Disconnection Cause
0	Unknown
1	Participant hung up
2	Disconnected by User
5	Resources deficiency
6	Password failure
20	H323 call close. No port left for audio
21	H323 call close. No port left for video
22	H323 call close. No port left for FECC
23	H323 call close. No control port left
25	H323 call close. No port left for video content
51	A common key exchange algorithm could not be established between the MCU and the remote device
53	Remote device did not open the encryption signaling channel
59	The remote devices' selected encryption algorithm does not match the local selected encryption algorithm
141	Called party not registered
145	Caller not registered
152	H323 call close. ARQ timeout
153	H323 call close. DRQ timeout
154	H323 call close. Alt Gatekeeper failure

Table C-35 Disconnection Cause Values (Continued)

Value	Call Disconnection Cause
191	H323 call close. Remote busy
192	H323 call close. Normal
193	H323 call close. Remote reject
194	H323 call close. Remote unreachable
195	H323 call close. Unknown reason
198	H323 call close. Small bandwidth
199	H323 call close. Gatekeeper failure
200	H323 call close. Gatekeeper reject ARQ
201	H323 call close. No port left
202	H323 call close. Gatekeeper DRQ
203	H323 call close. No destination IP value
204	H323 call close. Remote has not sent capability
205	H323 call close. Audio channels not open
207	H323 call close. Bad remote cap
208	H323 call close. Capabilities not accepted by remote
209	H323 failure
210	H323 call close. Remote stop responding
213	H323 call close. Master slave problem
251	SIP timer popped out
252	SIP card rejected channels
253	SIP capabilities don’t match
254	SIP remote closed call
255	SIP remote cancelled call
256	SIP bad status

Table C-35 Disconnection Cause Values (Continued)

Value	Call Disconnection Cause
257	SIP remote stopped responding
258	SIP remote unreachable
259	SIP transport error
260	SIP bad name
261	SIP trans error TCP invite
300	SIP redirection 300
301	SIP moved permanently
302	SIP moved temporarily
305	SIP redirection 305
380	SIP redirection 380
400	SIP client error 400
401	SIP unauthorized
402	SIP client error 402
403	SIP forbidden
404	SIP not found
405	SIP client error 405
406	SIP client error 406
407	SIP client error 407
408	SIP request timeout
409	SIP client error 409
410	SIP gone
411	SIP client error 411
413	SIP client error 413
414	SIP client error 414

Table C-35 Disconnection Cause Values (Continued)

Value	Call Disconnection Cause
415	SIP unsupported media type
420	SIP client error 420
480	SIP temporarily not available
481	SIP client error 481
482	SIP client error 482
483	SIP client error 483
484	SIP client error 484
485	SIP client error 485
486	SIP busy here
487	SIP request terminated
488	SIP client error 488
500	SIP server error 500
501	SIP server error 501
502	SIP server error 502
503	SIP server error 503
504	SIP server error 504
505	SIP server error 505
600	SIP busy everywhere
603	SIP global failure 603
604	SIP global failure 604
606	SIP global failure 606

MGC Manager Events that are not Supported by the RMX 2000

The following MGC Manager events are not supported by the RMX 2000:

For details of these events see the M*GC Manager User's Guide Volume II*, *Appendix A*.

- Event 8 - REMOTE COM MODE
- Event 11 - ATM CHANNEL CONNECTED
- Event 12 - ATM CHANNEL DISCONNECTED
- Event 13 - MPI CHANNEL CONNECTED
- Event 14 - MPI CHANNEL DISCONNECTED
- Event 15 - H323 CALL SETUP
- Event 16 - H323 CLEAR INDICATION
- Event 24 - SIP CALL SETUP
- Event 25 - SIP CLEAR INDICATION
- Event 27 - RECORDING SYSTEM LINK
- Event 110 - OPERATOR ON HOLD PARTY
- Event 113 - CONFERENCE REMARKS
- Event 2108 - OPERATOR MOVE PARTY TO CONFERENCE CONTINUE 1
- Event 3001 - CONFERENCE START CONTINUE 2
- Event 3108 - OPERATOR MOVE PARTY TO CONFERENCE CONTINUE 2
- Event 4001 - CONFERENCE START CONTINUE 3
- Event 4108 - OPERATOR MOVE PARTY TO CONFERENCE CONTINUE 3

Appendix D

Ad Hoc Conferencing and External Database Authentication

The RMX Ad Hoc conferencing feature enables participants to start ongoing conferences on-the-fly, without prior definition when dialing an Ad Hoc-enabled Entry Queue. The created conference parameters are taken from the Profile assigned to the Ad Hoc-enabled Entry Queue.

Ad Hoc conferencing is available in two modes:

- ***Ad Hoc Conferencing without Authentication***

  Any participant can dial into an Entry Queue and initiate a new conference if the conference does not exist. This mode is usually used for the organization's internal Ad Hoc conferencing.
- Ad Hoc conferencing with external database authentication

  In this mode, the participant's right to start a new conference is validated against a database.

The external database application can also be used to validate the participant's right to join an ongoing conference. Conference access authentication can be:

- Part of the Ad Hoc conferencing flow where the participants must be authorized before they can enter the conference created in the Ad Hoc flow.
- Independent of Ad Hoc conferencing where conference access is validated for all conferences running on the MCU regardless of the method in which the conference was started.

Ad Hoc Conferencing without Authentication

A participant dials in to an Ad Hoc-enabled Entry Queue and starts a new conference based on the Profile assigned to the Entry Queue. In this configuration, any participant connecting to the Entry Queue can start a new conference, and no security mechanism is applied. This mode is usually used in organizations where Ad Hoc conferences are started from within the network and without security breach.

Starting a conference uses the following method:

1. The participant dials in to the Ad Hoc-enabled Entry Queue.
2. The Conference ID is requested by the system.
3. The participant inputs a Conference ID via his/her endpoint remote control using DTMF codes.
4. The MCU checks whether a conference with the same Conference ID is running on the MCU. If there is such a conference, the participant is moved to that conference. If there is no ongoing conference with that Conference ID, the system creates a new conference, based on the Profile assigned to the Entry Queue, and connects this participant as the conference chairperson.

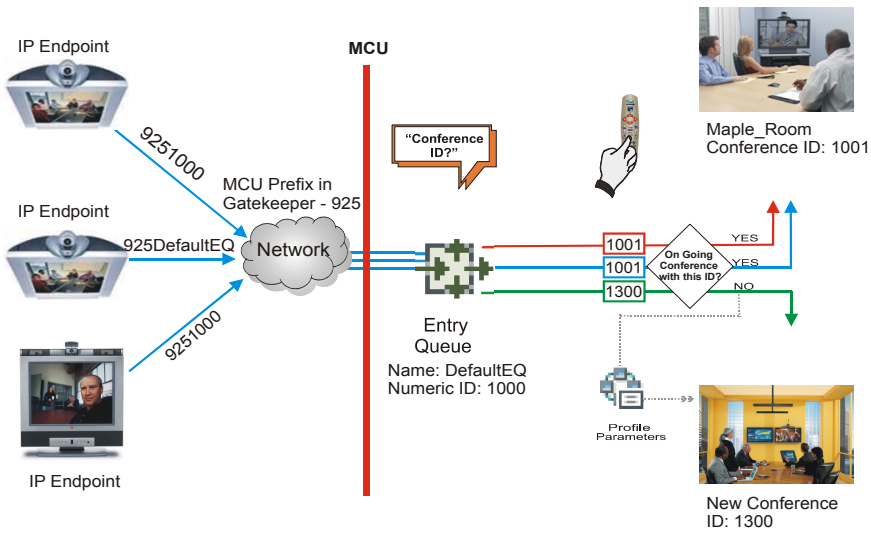


Figure D-1 Ad Hoc Conference Initiation without Authentication

To enable this workflow, the following components must be defined in the system:

- An Entry Queue IVR Service with the appropriate audio file requesting the Conference ID
- An Ad Hoc-enabled Entry Queue with an assigned Profile

Ad Hoc Conferencing with Authentication

The MCU can work with an external database application to validate the participant's right to start a new conference. The external database contains a list of participants, with their assigned parameters. The conference ID entered by the participant is compared against the database. If the system finds a match, the participant is granted the permission to start a new conference.

To work with an external database application to validate the participant's right to start a new conference, the Entry Queue IVR Service must be configured to use the external database application for authentication. In the external database application, you must define all participants (users) with rights to start a new conference using Ad Hoc conferencing. For each user defined in the database, you enter the conference ID, Conference Password (optional) and Chairperson Password (when applicable), billing code, Conference general information (corresponding to the User Defined 1 field in the Profile properties) and user's PIN code. The same user definitions can be used for conference access authentication, that is, to determine who can join the conference as a participant and who as a chairperson.

Entry Queue Level - Conference Initiation Validation with an External Database Application

Starting a new conference with external database application validation entails the following steps:

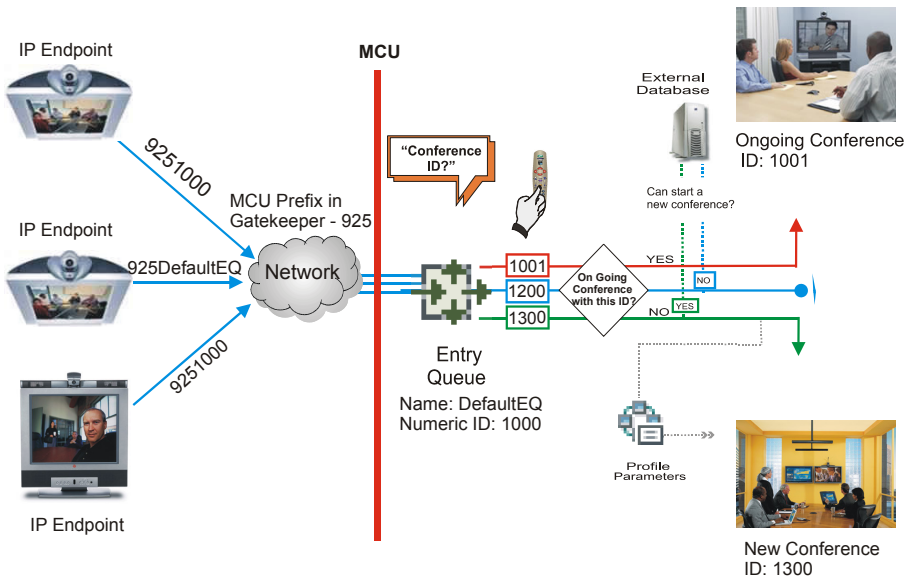


Figure D-2 Conference Initiation Validation with External Database Application

1. The participant dials in to an Ad Hoc-enabled Entry Queue.
2. The participant is requested to enter the Conference ID.
3. The participant enters the conference ID via his/her endpoint remote control using DTMF codes. If there is an ongoing conference with this Conference ID, the participant is moved to that conference where another authentication process can occur, depending on the IVR Service configuration.
4. If there is no ongoing conference with that Conference ID, the MCU verifies the Conference ID with the database application that compares it against its database. If the database application finds a match, the external database application sends a response back to the MCU, granting the participant the right to start a new ongoing conference.

If this Conference ID is not registered in the database, the conference cannot be started and this participant is disconnected from the Entry Queue.

5 The external database contains a list of participants (users), with their assigned parameters. Once a participant is identified in the database (according to the conference ID), his/her parameters (as defined in the database) can be sent to the MCU in the same response granting the participant the right to start a new ongoing conference. These parameters are:

- Conference Name
- Conference Billing code
- Conference Password
- Chairperson Password
- Conference Information, such as the contact person name. These fields correspond to Info 1, 2 and 3 fields in the *Conference Properties - Information* dialog box.
- Maximum number of participants allowed for the conference
- Conference Owner

These parameters can also be defined in the conference Profile. In such a case, parameters sent from the database overwrite the parameters defined in the Profile. If these parameters are not sent from the external database to the MCU, they will be taken from the Profile.

6 A new conference is started based on the Profile assigned to the Entry Queue.

7 The participant is moved to the conference.
If no password request is configured in the Conference IVR Service assigned to the conference, the participant that initiated the conference is directly connected to the conference, as its chairperson.

If the Conference IVR Service assigned to the conference is configured to prompt for the conference password and chairperson password, without external database authentication, the participant has to enter these passwords in order to join the conference.

To enable this workflow, the following components must be defined in the system:

- A Conference IVR Service with the appropriate prompts. If conference access is also validated with the external database

application it must be configured to access the external database for authentication.

- An Entry Queue IVR Service configured with the appropriate audio prompts requesting the Conference ID and configured to access the external database for authentication.
- Create a Profile with the appropriate conference parameters and the appropriate Conference IVR Service assigned to it.
- An Ad Hoc-enabled Entry Queue with the appropriate Entry Queue IVR Service and Conference Profile assigned to it.
- An external database application with a database containing Conference IDs associated with participants and their relevant properties.
- Define the flags required to access the external database in System Configuration. For more information, see "***MCU Configuration to Communicate with an External Database Application"*** on page D-13.

Conference Access with External Database Authentication

The MCU can work with an external database application to validate the participant's right to join an existing conference. The external database contains a list of participants, with their assigned parameters. The conference password or chairperson password entered by the participant is compared against the database. If the system finds a match, the participant is granted the permission to access the conference.

To work with an external database application to validate the participant's right to join the conference, the Conference IVR Service must be configured to use the external database application for authentication.

Conference access authentication can be performed as:

- Part of the Ad Hoc conferencing flow where the participants must be authorized before they can enter the conference created in the Ad Hoc flow
- Independent of Ad Hoc conferencing where conference access is validated for all conferences running on the MCU regardless of the method in which the conference was started.

Conference access authentication can be implemented for all participants joining the conference or for chairpersons only.

Conference Access Validation - All Participants (Always)

Once the conference is created either via an Ad Hoc Entry Queue, or a standard ongoing conference, the right to join the conference is authenticated with the external database application for all participants connecting to the conference.

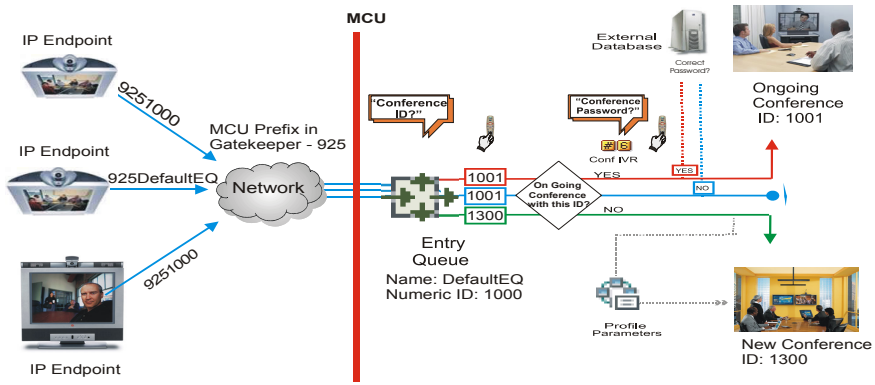


Figure D-3 *Conference Access - Conference Password validation with External Database Application*

Joining the conference entails the following steps:

- When the conference is started (either in the Ad Hoc flow or in the standard method), all participants connecting to the conference are moved to the Conference IVR queue where they are prompted for the conference password.
- When the participant enters the conference password or his/her personal password, it is sent to the external database application for validation.
- If there is a match, the participant is granted the right to join the conference. In addition, the external database application sends to the MCU the following parameters:
  - Participant name (display name)
  - Whether or not the participant is the conference chairperson
  - Participant Information, such as the participant E-mail. These fields correspond to Info 1, 2, 3 and 4 fields in the *Participant Properties - Information* dialog box.

If there is no match (i.e. the conference or personal password are not defined in the database), the request to access the conference is rejected and the participant is disconnected from the MCU.

- If the Conference IVR Service is configured to prompt for the chairperson identifier and password, the participant is requested to enter the chairperson identifier.
  - — If no identifier is entered, the participant connects as a standard, undefined participant.
- If the chairperson identifier is entered, the participant is requested to enter the chairperson password. In this flow, the chairperson password is **not** validated with the external database application, only with the MCU.
  - — If the correct chairperson password is entered, the participant is connected to the conference as its chairperson.
  - — If the wrong password is entered, he/she is disconnected from the conference.

To enable conference access validation for all participants the following conferencing components are required:

- The external database must hold the conference password or the participant personal password/PIN code or the participant's Alias.
- The Conference IVR Service assigned to the conference (defined in the Profile) must be configured to authenticate the participant's right to access the conference with the external database application for all requests. In addition it must be configured to prompt for the Conference Password.

Conference Access Validation - Chairperson Only (Upon Request)

An alternative validation method at the conference level is checking only the chairperson password with the external database application. All other participants can be checked only with the MCU (if the Conference IVR Service is configured to prompt for the conference password) or not checked at all (if the Conference IVR Service is configured to prompt only for the chairperson password).

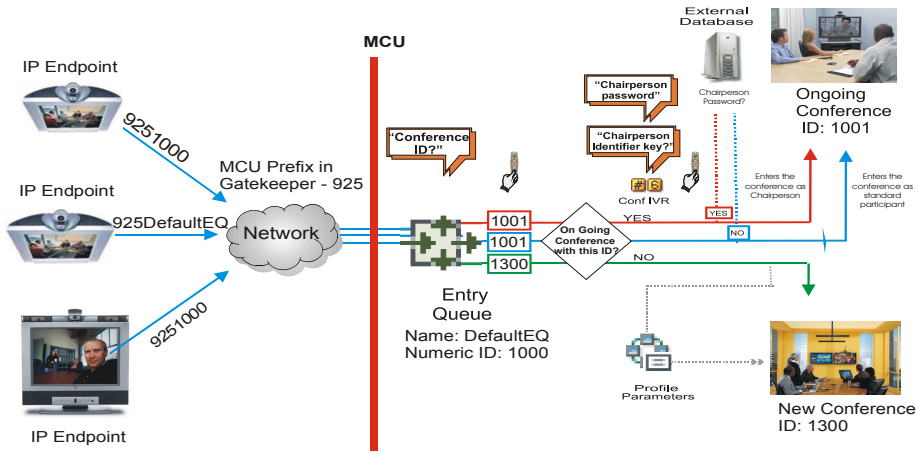


Figure D-4 *Conference Access - Chairperson Password validation with external database application*

Joining the conference entails the following steps:

- When the conference is started (either in the Ad Hoc flow or in the standard method), all participants connecting to the conference are moved to the conference IVR queue where they are prompted for the conference password.
- If the Conference IVR Service is configured to prompt for the Conference password, the participant is requested to enter the conference password. In this flow, the conference password is **not** validated with the external database application, only with the MCU.
  - If the wrong password is entered, he/she is disconnected from the conference.

- If the correct conference password is entered, the participant is prompted to enter the chairperson identifier key.
  - If no identifier is entered, the participant is connected to the conference as a standard participant.
- If the chairperson identifier is entered, the participant is prompted to enter the chairperson password.
- When the participant enters the chairperson password or his/her personal password, it is sent to the external database application for validation.
  - If the password is incorrect the participant is disconnected from the MCU.
- If there is a match, the participant is granted the right to join the conference as chairperson. In addition, the external database application sends to the MCU the following parameters:
  - Participant name (display name)
  - Participant Information, such as the participant E-mail. These fields correspond to Info 1, 2, 3 and 4 fields in the *Participant Properties - Information* dialog box.

To enable conference access validation for all participants the following conferencing components are required:

- The external database must hold the Chairperson Password or the participant's Alias.
- The Conference IVR Service assigned to the conference (defined in the Profile) must be configured to check the external database for the Chairperson password only when the participant enters the chairperson identifier key (either pound or star). In addition, it must be configured to prompt for the chairperson identifier key and password.

System Settings for Ad Hoc Conferencing and External Database Authentication

Ad Hoc Settings

Before a participant can initiate an Ad Hoc conference (with or without authentication), the following components must be defined:

- **Profiles**

  Defines the conference parameters for the conferences that will be initiated from the Ad Hoc-enabled Entry Queue. For more information, see "***Conference Profiles"*** on page 1-1.

- **Entry Queue IVR Service with Conference ID Request Enabled**

  The Entry Queue Service is used to route participants to their destination conferences, or create a new conference with this ID.

  In Ad Hoc conferencing, the Conference ID is used to check whether the destination conference is already running on the MCU and if not, to start a new conference using this ID. For more information, see "***Entry Queues IVR Service"*** on page 13-26.

- **Ad Hoc - enabled Entry Queue**

  Ad Hoc conferencing must be enabled in the Entry Queue and a Profile must be assigned to the Entry Queue. In addition, an Entry Queue IVR Service supporting conference ID request. For more information, see "***Ad Hoc Conferencing"*** on page 4-10.

Authentication Settings

- **MCU Configuration**

  Usage of an external database application for authentication (for starting new conferences or joining ongoing conferences) is configured for the MCU in the System Configuration.

- **Entry Queue IVR Service with Conference Initiation Authentication Enabled**

  Set the Entry Queue IVR Service to send authentication requests to the external database application to verify the participant's right to start a new conference according to the Conference ID entered by the participant.

- **Conference IVR Service with Conference Access Authentication Enabled**

  Set the Conference IVR Service to send authentication requests to the external database application to verify the participant's right to connect to the conference as a standard participant or as a chairperson.

- **External Database Application Settings**

  The external database contains a list of participants (users), with their assigned parameters. These parameters are:

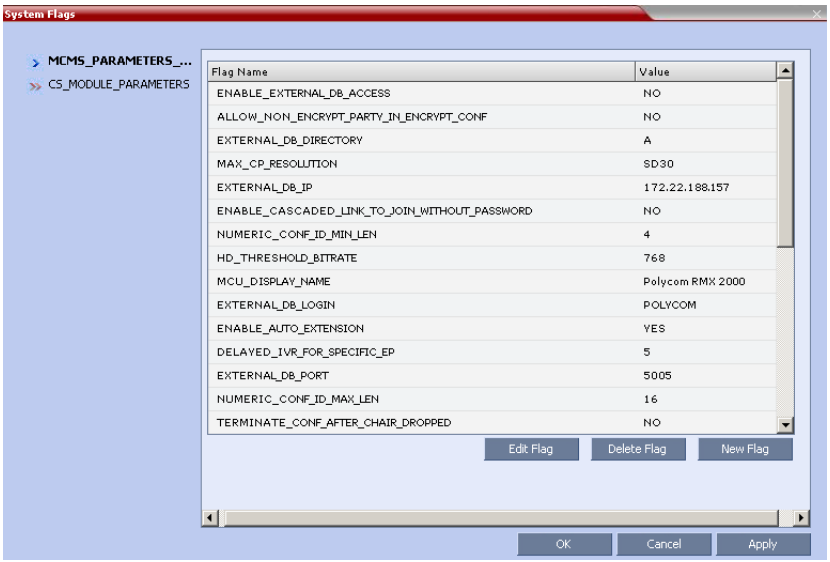
  — Conference Name
  — Conference Billing code
  — Conference Password
  — Chairperson Password
  — Conference Information, such as the contact person name. These fields correspond to Info 1, 2 and 3 fields in the *Conference Properties - Information* dialog box.
  — Maximum number of participants allowed for the conference
  — Conference Owner
  — Participant name (display name)
  — Participant Information, such as the participant E-mail. These fields correspond to Info 1, 2, 3 and 4 fields in the *Participant Properties - Information* dialog box.

MCU Configuration to Communicate with an External Database Application

To enable the communication with the external database application, several flags must be set in the System Configuration.

To set the System Configuration flags:

1 On the *Setup* menu, click **System Configuration**.
The *System Flags* dialog box opens.



2 Modify the values of the following flags:

Table D-1 *Flag Values for Accessing External Database Application*

Flag	Description and Value
ENABLE_EXTERNAL_DB_ACCESS	The flag that enables the use of the external database application.
EXTERNAL_DB_IP	The IP address of the external database application server. default IP: 0.0.0.0.

Table D-1 *Flag Values for Accessing External Database Application (Continued)*

Flag	Description and Value
EXTERNAL_DB_ PORT	The port number used by the MCU to access the external application server. Default Port = 80. To use the WebCommander application as an external database application, you must specify 5005.
EXTERNAL_DB_ LOGIN	The user name defined in the external database application for the MCU. To use the WebCommander application as an external database application, the default user name is POLYCOM.
EXTERNAL_DB_ PASSWORD	The password associated with the user name defined for the MCU in the external database application. To use the WebCommander application as an external database application, the default password is POLYCOM.
EXTERNAL_DB_ DIRECTORY	The URL of the external database application.

For more information about flag settings, see "***System Configuration"*** on page **16-10**.

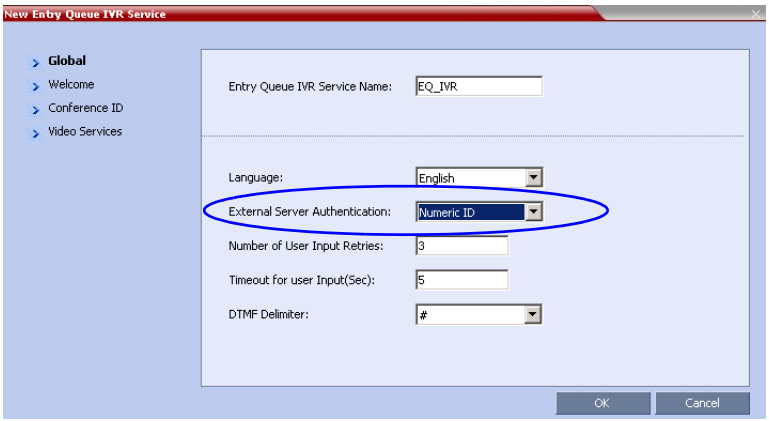
3 Click **OK**.

4 Reset the MCU for flag changes to take effect.

Enabling External Database Validation for Starting New Ongoing Conferences

The validation of the participant's right to start a new conference with an external database application is configured in the *Entry Queue IVR Service - Global* dialog box.

- Set the *External Server Authentication* field to **Numeric ID**.

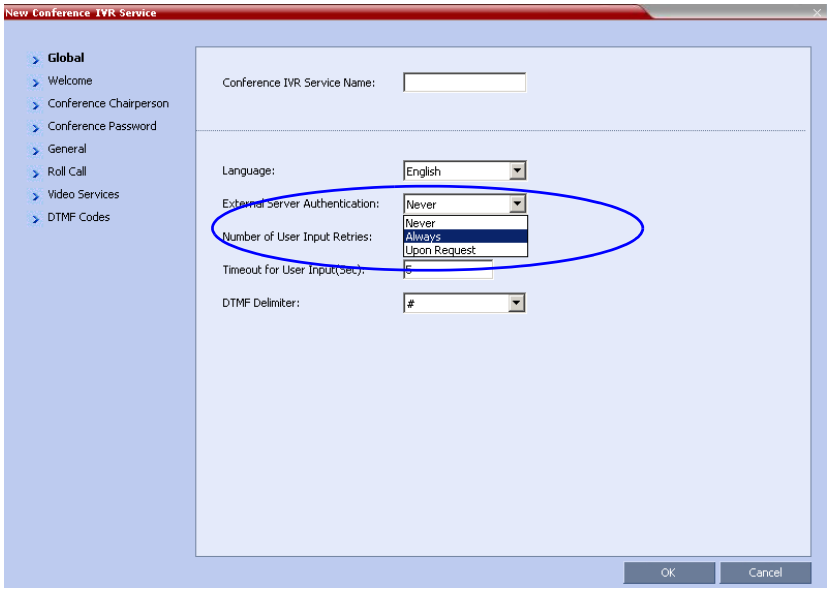


Enabling External Database Validation for Conferences Access

The validation of the participant's right to join an ongoing conference with an external database application is configured in the *Conference IVR Service - Global* dialog box.

You can set the system to validate all the participants joining the conference or just the chairperson.

- Set the *External Server Authentication* field to:
  - **Always -** to validate the participant's right to join an ongoing conference for all participants
  - **Upon Request** - to validate the participant's right to join an ongoing conference as chairperson



Appendix E

Participant Properties Advanced Channel Information

The following appendix details the properties connected with information about audio and video parameters, as well as, problems with the network which can affect the audio and video quality.

Table E-1 *Participant Properties - Channel Status Advanced Parameters*

Field	Description
Media Info	
Algorithm	Indicates the audio or video algorithm and protocol.
Frame per packet (audio only)	The number of audio frames per packet that are transferred between the MCU and the endpoint. If the actual Frame per Packets are higher than Frame per Packets declared during the capabilities exchange, a Faulty flag is displayed.
Resolution (video only)	Indicates the video resolution in use. If the actual resolution is higher than resolution declared in the capabilities exchange, the Faulty flag is displayed. For example, if the declared resolution is CIF and the actual resolution is 4CIF, the Faulty flag is displayed.

Table E-1 *Participant Properties - Channel Status Advanced Parameters*

Field	Description
Frame Rate (video only)	The number of video frames per second that are transferred between the MCU and the endpoint.
Annexes (video only)	Indicates the H.263 annexes in use at the time of the last RTCP report. If the actual annexes used are other than the declared annexes in the capabilities exchange, the Faulty flag is displayed.
Channel Index	For Polycom Internal use only.
RTP Statistics	
Actual loss	The number of missing packets counted by the IP card as reported in the last RTP Statistics report. If a packet that was considered lost arrives later, it is deducted from the packet loss count. Packet loss is displayed with the following details: • ***Accumulated N*** - number of lost packets accumulated since the channel opened. • ***Accumulated %*** - percentage of lost packets out of the total number of packets transmitted since the channel opened. • ***Interval N*** - number of packets lost in the last RTP report interval (default interval is 5 minutes). • ***Interval %*** - percentage of lost packets out of the total number of packets transmitted in the last RTP report interval (default interval is 5 minutes). • ***Peak*** - the highest number of lost packets in a report interval from the beginning of the channel's life span.

Table E-1 *Participant Properties - Channel Status Advanced Parameters*

Field	Description
Out of Order	The number of packets arriving out of order. The following details are displayed: • ***Accumulated N*** - total number of packets that arrived out of order since the channel opened. • ***Accumulated %*** - percentage of packets that arrived out of order out of the total number of packets transmitted since the channel opened. • ***Interval N*** - number of packets that arrived out of order in the last RTP report interval (default interval is 5 minutes). • ***Interval %*** - percentage of packets that arrived out of order out of the total number of packets transmitted in the last RTP report interval (default interval is 5 minutes). • ***Peak*** - the highest number of packets that arrived out of order in a report interval from the beginning of the channel's life span.

Table E-1 *Participant Properties - Channel Status Advanced Parameters*

Field	Description
Fragmented	Indicates the number of packets that arrived to the IP card fragmented (i.e., a single packet broken by the network into multiple packets). This value can indicate the delay and reordering of fragmented packets that require additional processing, but is not considered a fault. The Fragmented information is displayed with the following details: • ***Accumulated N*** - total number of packets that were fragmented since the channel opened. • ***Accumulated %*** - percentage of fragmented packets out of the total number of packets transmitted since the channel opened. • ***Interval N*** - number of fragmented packets received in the last RTP report interval (default interval is 5 minutes). • ***Interval %*** - percentage of fragmented packets out of the total number of packets transmitted in the last RTP report interval (default interval is 5 minutes). • ***Peak*** - the highest number of fragmented packets in a report interval from the beginning of the channel's life span.

Appendix F

Secure Communication Mode

The RMX can be configured to work in *Secure Mode* by configuring the *RMX* and the *RMX Web Client* to work with SSL/TLS.

In this mode, a SSL/TLS Certificate is installed on the MCU, setting the MCU Listening Port to secured port 443.

TLS is a cryptographic protocol used to ensure secure communications on public networks. TLS uses a *Certificate* purchased from a trusted third party *Certificate Authority* to authenticate public keys that are used in conjunction with private keys to ensure secure communications across the network.

The RMX supports:

- TLS 1.0
- SSL 3.0 (Secure Socket Layer)

Both TLS 1.0 and SSL 3.0 utilize 1024-bit RSA public key encryption.

Switching to Secure Mode

The following operations are required to switch the *RMX* to *Secure Mode*:

- Purchase and Install the *SSL/TLS certificate*
- Modify the *Management Network* settings
- Create/Modify the relevant *System Flags*

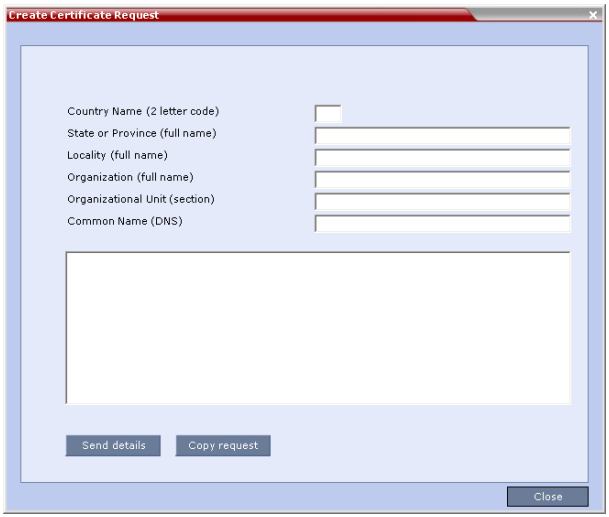
Purchasing a Certificate

Once a certificate is purchased and received it is stored in the RMX and used for all subsequent secured connections.

To create/purchase a certificate:

1 In the *RMX* menu, click **Setup > Secured RMX Communications > Create certificate request**.

The *Create Certificate Request* dialog box is displayed.



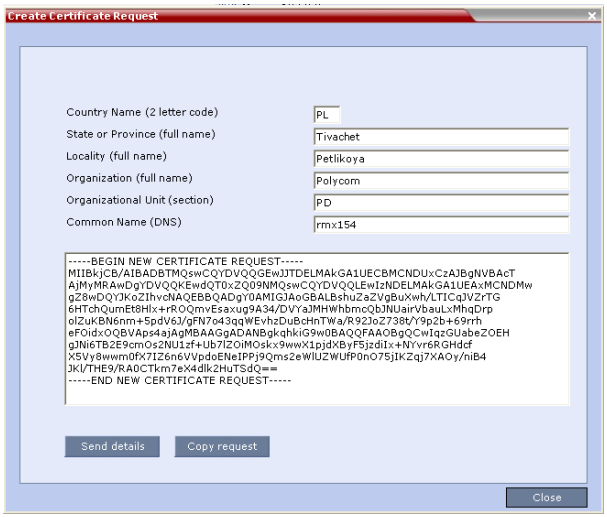
2 Enter information in all the following fields:

Table F-1 *Create Certificate Request*

Field	Description
Country Name	Enter any 2 letter code for the country name.
State or Province	Enter the full name of the state or province.
Locality	Enter the full name of the town/city/location.
Organization	Enter the full name of your organization for which the certificate will be issued.
Organizational Unit	Enter the full name of the unit (group or division) for which the certificate will be issued.
Common Name (DNS/ IP)	Enter the *DNS MCU Host Name*. This *MCU Host Name* must also be configured in the *Management Network Properties* dialog box.

3 Click **Send Details**.

The RMX creates a *New Certificate Request* and returns it to the *Create Certificate Request* dialog box along with the information the user submitted.



4 Click **Copy Request** to copy the *New Certificate Request* to the workstation's clipboard.

5 Connect to your preferred *Certificate Authority's* website using the web browser.

6 Follow the purchasing instructions at the *Certificate Authority's* website.
Paste **(Ctrl + V)** the *New Certificate Request* as required by the *Certificate Authority*.

The *Certificate Authority* issues the TLS/SSL certificate, and sends the certificate to you by e-mail.

Installing the Certificate

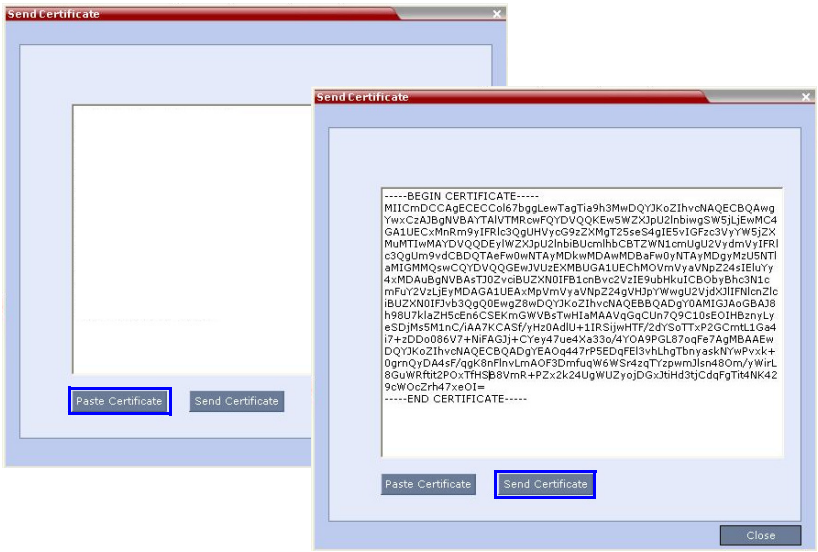
To install the certificate:

After you have received the certificate from the *Certificate Authority:*

1 **Copy** (**Ctrl + C**) the certificate information from the *Certificate Authority's* e-mail to the clipboard.

2 In the *RMX* menu, click **Setup > Secured RMX Communications > Send Certificate**.

3 Click **Paste Certificate** to paste the clipboard content into the *Send Certificate* dialog box.



4 Click the **Send Certificate** button to send the certificate to the RMX.

The MCU validates the certificate.

— If the certificate is not valid, an error message is displayed.

— If the certificate matches the private key, and the task is completed, a confirmation message indicating that the certificate was created successfully is displayed.

A *System Restart* is **not** required at this point.

The certificate expiry date is checked daily. An active alarm is raised two weeks before the certificate is due to expire, stating the number of days to expiry.

If the certificate expires, the RMX continues to work in secure mode and an *Active Alarm* is raised with *Security mode failed – Certificate expired* in the description field.

Certificates are deleted when an administrator performs a *Restore Factory Defaults* with the *Comprehensive Restore* option selected.

Creating/Modifying System Flags

The following *System Flags* in *system.cfg* control secure communications.

- *RMX_MANAGEMENT_SECURITY_PROTOCOL*
- *EXTERNAL_DB_PORT*

Appendix F, "System Flags" , below, lists both flags and their settings.

If the *System Flag*, *RMX_MANAGEMENT_SECURITY_PROTOCOL* does not exist in the system, it must be created by using the *RMX Setup* menu.

For more information see the *RMX 2000 Administrator's Guide*, "*Modifying System Flags"* on page **16-10**.

***Table F-2** System Flags*

Flag	Description
RMX_MANAGEMENT_ SECURITY_PROTOCOL	Enter the protocol to be used for secure communications. Default: TLSV1_SSLV3 (both). Default for U.S. Federal licenses: TLSV1.
EXTERNAL_DB_PORT	The external database server port used by the RMX to send and receive XML requests/ responses. For secure communications set the value to 443. Default: 5005.

The RMX must be restarted for modified flag settings to take effect.

Enabling Secure Communication Mode

After the SSL/TLS Certificate is installed, secure communications are enabled by modifying the properties of the *Management Network* in the *Management Network* properties dialog box.

When *Secure Communications Mode* is enabled:

- Only **`https://`** commands from the browser to the *Control Unit IP Address* of the RMX are accepted.
- The RMX listens only on secured port 443.
- All connection attempts on port 80 are rejected.

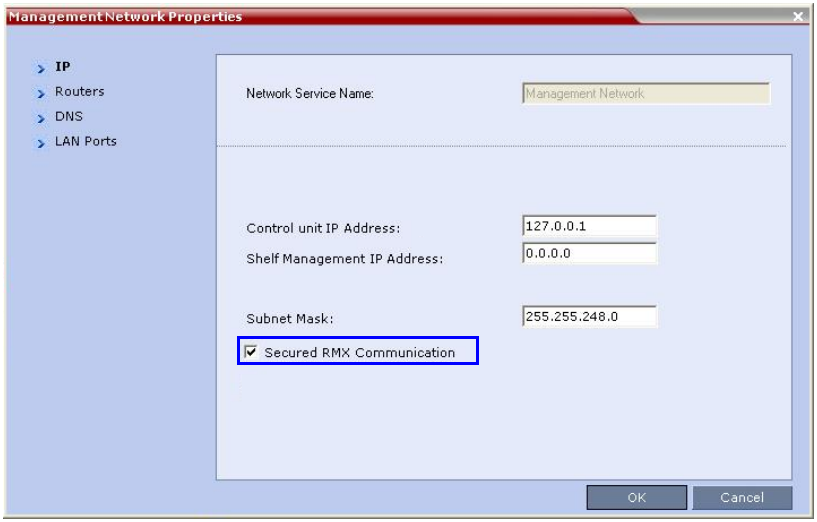
- A secure communication indicator ( ) is displayed in the browser's status bar.

To enable secure communications mode:

1 In the *RMX Management* pane, click **IP Network Services.**

2 In the *IP Network Services* list pane, double click the **Management Network** entry.

The *Management Network Properties* dialog box is displayed.



3 Select the *Secured RMX Communication* check box.

4 Click **OK**.

Alternate Management Network

The *Alternate Management Network* enables direct access to the RMX for support purposes. Access to the Alternate Management Network is via a cable connected to a workstation. The Alternate Management Network is accessible only via the dedicated LAN 3 port.

For more information see the *RMX 2000 Administrator's Guide*, "*Configuring Direct Connections to RMX*" on page **G-1** and "*Connecting to the Alternate Management Network*" on page **G-8**.

Connection to the *Alternate Management Network* bypasses LAN and Firewall security. Strict control of access to *LAN 3* port is recommended.

Securing an External Database

TLS 1.0 is used when securing communications between the RMX and an external database. The certificate is installed on the database server and the RMX is the client. When the certificate is installed on the database server, all client requests and responses are transferred via secure port 443.

It is important to verify that the external database application is operating in secure mode before enabling secure external database communications on the RMX. The RMX checks the validity of external database's certificate before communicating. If there is a certificate error an *Active Alarm* is raised with *Error in external database certificate* in the description field.

To enable secure RMX Communications with an External Database:

- Set the RMX to communicate with the database server via port 443 by setting the value of the *System Flag* `EXTERNAL_DB_PORT` in *system.cfg* to 443.

  For more information see the *RMX 2000 Administrator's Guide*, "*Modifying System Flags*" on page **16-10**.

Appendix G

Configuring Direct Connections to RMX

Direct connection to the RMX is necessary if you want to:

- Modify the RMX's *Factory Default Management Network* settings without using the USB key.
- Connect to the RMX's *Alternate Management Network* for support purposes.
- Connect to the RMX via a modem.

Management Network (Primary)

If you do not want to use the USB key method of modifying the RMX's *Management Network* parameters, it is necessary to establish a direct connection between a workstation and the RMX.

Alternate Management Network

The *Alternate Management Network* enables direct access to the RMX for support purposes.

While being separate from all other networks, it has identical functionality to the *Management Network*.

Support personnel can log in and used it to manage the RMX if a connection to the *Management Network* cannot be made or if internet access to the host network is blocked by LAN security or a firewall.

The *Alternate Management Network* cannot be configured and operates according to factory defaults.

The administrator's **Login** name, **Password**, viewing and system permissions on the *Alternate Management Network* are the same as those on the *Management Network*.

Once logged in, the *RMX Web Client* behaves as if the administrator had logged in on the *Management Network*.

Connection to the *Alternate Management Network* bypasses LAN and Firewall security. Strict control of access to *LAN 3* port is recommended.

Configuring the Workstation

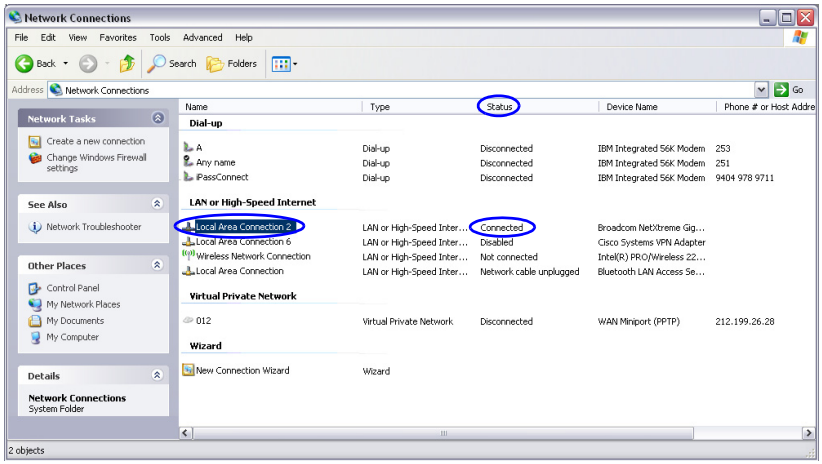
The following procedures show how to modify the workstation's networking parameters using the *Windows New Connection Wizard*.

For non-Windows operating systems an equivalent procedure must be performed by the system administrator.

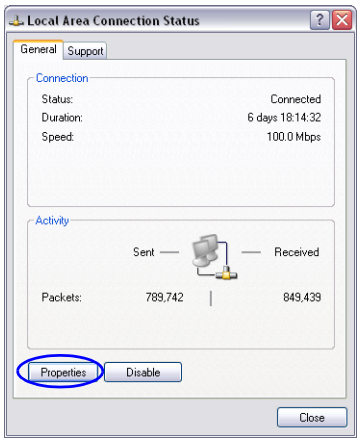
Before connecting directly, you must modify the *IP Address*, *Subnet Mask* and *Default Gateway* settings of the workstation to be compatible with either the RMX's *Default Management Network* or *Alternate Management Network*.

To modify the workstation's IP addresses:

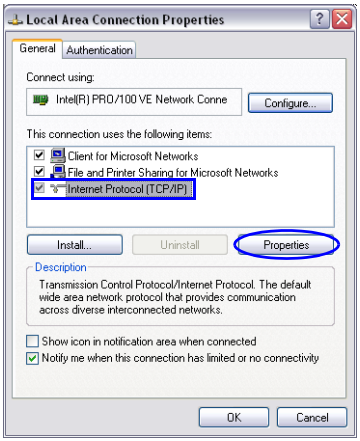
1. On the Windows *Start* menu, select **Settings > Network Connections.**
2. In the *Network Connections* window, double-click the **Local Area Connection** that has *Connected* status.


Network Connections
File Edit View Favorites Tools Advanced Help
Back Search Folders
Address Network Connections
Network Tasks
Create a new connection
Change Windows Firewall settings
See Also
Network Troubleshooter
Other Places
Control Panel
My Network Places
My Documents
My Computer
Details
Network Connections
System Folder
Name Type Status Device Name Phone # or Host Addre
Dial-up
A Dial-up Disconnected IBM Integrated 56K Modem 253
Any name Dial-up Disconnected IBM Integrated 56K Modem 251
iPassConnect Dial-up Disconnected IBM Integrated 56K Modem 9404 978 9711
LAN or High-Speed Internet
Local Area Connection 2 LAN or High-Speed Inter... Connected Broadcom NetXtreme Gig...
Local Area Connection 6 LAN or High-Speed Inter... Disabled Cisco Systems VPN Adapter
Wireless Network Connection LAN or High-Speed Inter... Not connected Intel(R) PRO/Wireless 22...
Local Area Connection LAN or High-Speed Inter... Network cable unplugged Bluetooth LAN Access Se...
Virtual Private Network
012 Virtual Private Network Disconnected WAN Miniport (PPTP) 212.199.26.28
Wizard
New Connection Wizard Wizard
2 objects

3 In the *Local Area Connection Status* dialog box, click the **Properties** button.

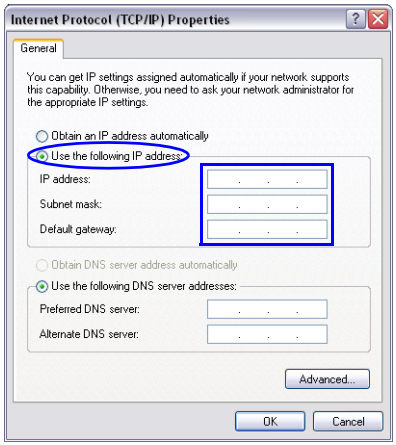


4 In the *Local Area Connection Properties* dialog box, select **Internet Protocol [TCP/IP] > Properties**.



5 In the *Internet Protocol (TCP/IP) Properties* dialog box, select **Use the following IP address**.

6 Enter the *IP address, Subnet mask* and *Default gateway* for the workstation.


Internet Protocol (TCP/IP) Properties
General
You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.
Obtain an IP address automatically
Use the following IP address:
IP address:
Subnet mask:
Default gateway:
Obtain DNS server address automatically
Use the following DNS server addresses:
Preferred DNS server:
Alternate DNS server:
Advanced...
OK
Cancel

The workstation's IP address should be in the same network neighborhood as the *RMX's Control Unit* IP address.

Example: *IP address* – near **`192.168.1.nn`**

None of the reserved IP addresses listed in *Table G-1* should be used for the IP Address.

The *Subnet mask* and *Default gateway* addresses should be the same as those for the RMX's *Management Network*.

The addresses needed for connection to either the RMX's *Default Management Network* or *Alternate Management Network* are listed in Table G-1.

For more information about connecting to the *Alternate Management Network*, see "*Connecting to the Alternate Management Network*" on page **G-8**.

Table G-1 *Reserved IP Addresses*

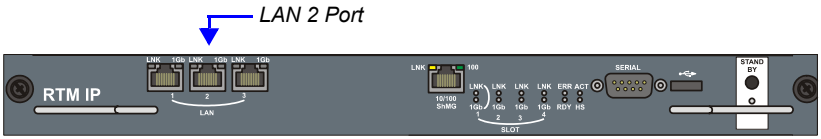
Network Entity	IP Address	
	Management Network (Factory Default)	Alternate Network
Control Unit IP Address	192.168.1.254	169.254.192.10
Control Unit Subnet Mask	255.255.255.0	255.255.240.0
Default Router IP Address	192.168.1.1	169.254.192.1
Shelf Management IP Address	192.168.1.252	169.254.192.16
Shelf Management Subnet Mask	255.255.255.0	255.255.240.0
Shelf Management Default Gateway	192.168.1.1	169.254.192.1

7 Click the **OK** button.

Connecting to the Management Network

To connect directly to the RMX:

1 Using a LAN cable, connect the workstation to the LAN 2 Port on the RMX's back panel.



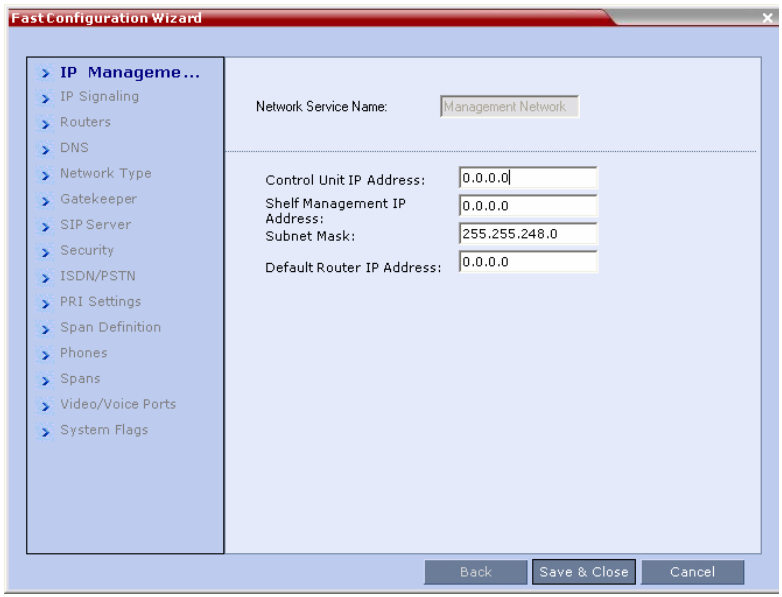
2 Connect the power cable and power the RMX **On**.

3 Start the *RMX Web Client* application on the workstation, by entering the factory setting *Management IP* address in the browser's address line and pressing **Enter**.

4 In the *RMX Web Client* Login screen, enter the default *Username* (**POLYCOM**) and *Password* (**POLYCOM**) and click the **Login** button.

The *Fast Configuration Wizard* starts.

If no *USB key* is detected and **either**: this is the *First Time Power-up* **or** the *Default IP Service* has been deleted and the RMX has been reset, the following dialog box is displayed:



For more information about First-time Power-up and the *Fast Configuration Wizard* see the *RMX 2000 Getting Started Guide*, *"Procedure 3: First-time Power-up and Connection to MCU"* on page **2-9**.

5 Enter the following parameters using the information supplied by your network administrator:

— *Control Unit IP Address*
— *Shelf Management IP Address*
— *Control Unit Subnet Mask*
— *Default Router IP Address*

6 Click the **Save & Close** button.

The system prompts you to sign in with the new *Control Unit IP Address*.

7 Disconnect the LAN cable between the workstation and the LAN 2 Port on the RMX's back panel.

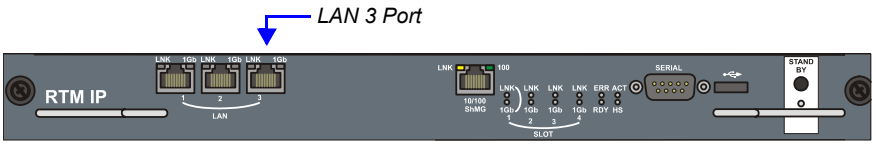
8 Connect LAN 2 Port on the RMX's back panel to the local network using a LAN cable.

9 Enter the new *Control Unit IP Address* in the browser's address line, using a workstation on the local network, and press **Enter** to start the *RMX Web Client* application.

10 In the *RMX Web Client* Login screen, enter the default *Username* (**POLYCOM**) and *Password* (**POLYCOM**) and click the **Login** button.

Connecting to the Alternate Management Network

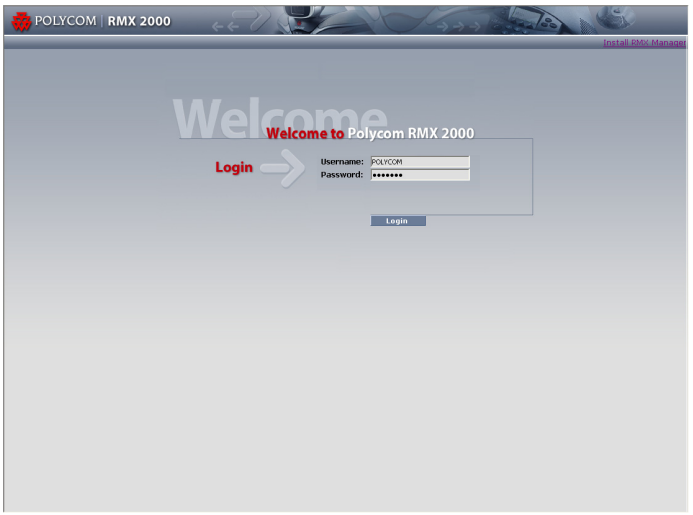
Access to the *Alternate Management Network* is via a cable connected to a workstation. The *Alternate Management Network* is accessible only via the dedicated *LAN 3* port.



To connect to the Alternate Management Network:

1. Connect the cable between the RMX's LAN 3 port and the LAN port configured on the workstation.
2. Start the *RMX Web Client* application on the workstation, by entering `http://169.254.192.10` (the *Control Unit IP Address*) in the browser's address line and pressing **Enter**.

   The *Login* dialog box is displayed.



3. In the *RMX Welcome Screen*, enter the administrator's *Username* and *Password* and click the **Login** button.

   The *RMX Web Client* starts and the RMX can be managed in the same manner as if you had logged on the *Management Network*.

Connecting to the RMX via Modem

Remote access to the RMX's *Alternate Management Network* is supported via an external PSTN <=> IP modem.

To connect via modem to the *Alternate Management Network* the following procedures must be performed:

1. **Procedure 1: Install the RMX Manager –** the web client enables direct access to the RMX for support purposes.
2. **Procedure 2: Configure the modem** – by assigning it an IP address on a specific subnet in the *Alternate Management Network*.
3. **Procedure 3: Create a dial-up connection –** using the *Windows New Connection Wizard.*
4. **Procedure 4: Connect to the RMX –** via the *RMX Manager***.**

Procedure 1: Install the RMX Manager

Before installing the *RMX Manager,* verify that you have at least 150Mb of free space on your workstation.

For more information see "*Installing RMX Manager"* on page **16-1**.

Procedure 2: Configure the Modem

Configure the modem as follows:

- **IP address –** near `169.254.192.nn`
- **Subnet Mask –** `255.255.240.0`

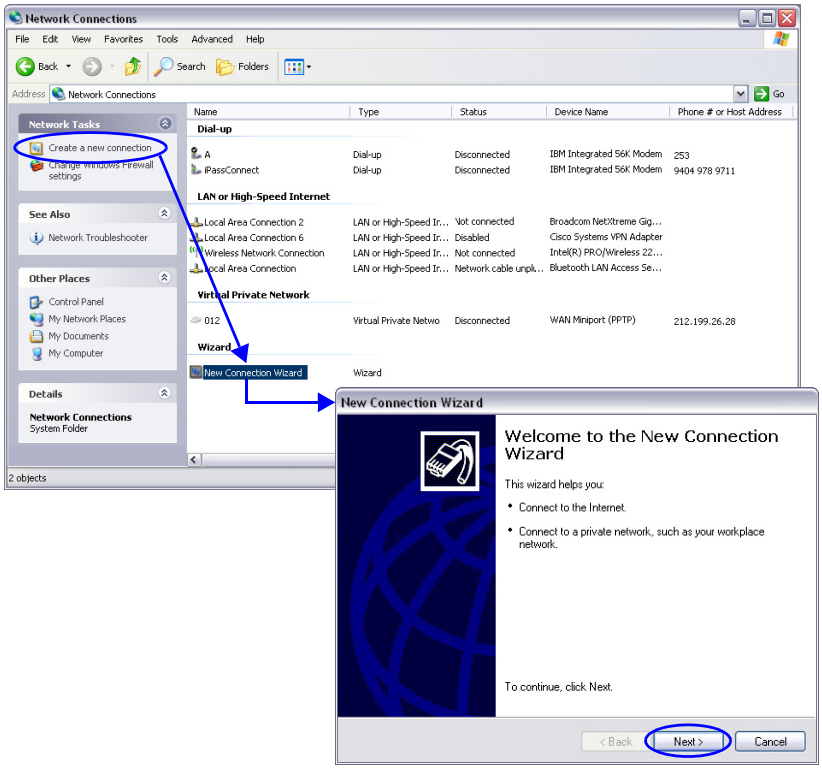
None of the reserved IP addresses listed in Table G-1 on page **G-5** should be used for the IP Address.

Procedure 3: Create a Dial-up Connection

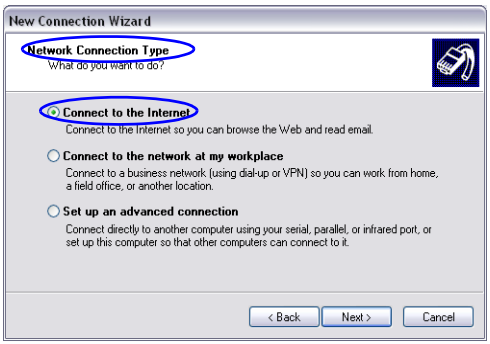
To create a dial-up connection:

This procedure is performed once. Only the *Dial* field in the *Connect* applet (see step 10 on page G-14) is modified for connection to different modems.

1. In *Windows*, navigate via the *Control Panel* to the *Network Connections* applet and select **Create a new connection.**
2. When the *New Connection Wizard* is displayed, click the **Next** button.


Network Connections
File Edit View Favorites Tools Advanced Help
Back Search Folders
Address Network Connections
Network Tasks
Create a new connection
Change Windows Firewall settings
See Also
Network Troubleshooter
Other Places
Control Panel
My Network Places
My Documents
My Computer
Details
Network Connections
System Folder
Name Type Status Device Name Phone # or Host Address
Dial-up
A Dial-up Disconnected IBM Integrated 56K Modem 253
iPassConnect Dial-up Disconnected IBM Integrated 56K Modem 9404 978 9711
LAN or High-Speed Internet
Local Area Connection 2 LAN or High-Speed Ir... Not connected Broadcom NetXtreme Gig...
Local Area Connection 6 LAN or High-Speed Ir... Disabled Cisco Systems VPN Adapter
Wireless Network Connection LAN or High-Speed Ir... Not connected Intel(R) PRO/Wireless 22...
Local Area Connection LAN or High-Speed Ir... Network cable unpl... Bluetooth LAN Access Se...
Virtual Private Network
012 Virtual Private Netwo Disconnected WAN Miniport (PPTP) 212.199.26.28
Wizard
New Connection Wizard Wizard
2 objects
New Connection Wizard
Welcome to the New Connection Wizard
This wizard helps you:
Connect to the Internet.
Connect to a private network, such as your workplace network.
To continue, click Next.
< Back Next > Cancel

3 In the *Network Connection Type* box, select **Connect to the Internet** and click the **Next** button.


New Connection Wizard
Network Connection Type
What do you want to do?
Connect to the Internet
Connect to the Internet so you can browse the Web and read email.
Connect to the network at my workplace
Connect to a business network (using dial-up or VPN) so you can work from home, a field office, or another location.
Set up an advanced connection
Connect directly to another computer using your serial, parallel, or infrared port, or set up this computer so that other computers can connect to it.
< Back
Next >
Cancel

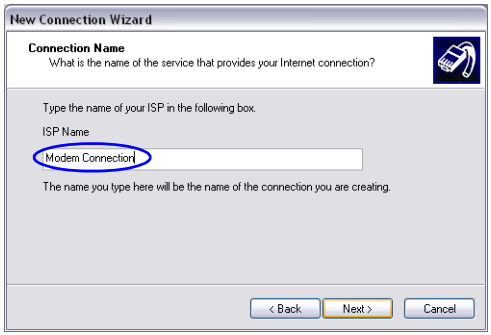
4 In the *Getting Ready* box, select **Set up my connection manually** and click the **Next** button.

New Connection Wizard
Getting Ready
The wizard is preparing to set up your Internet connection.
How do you want to connect to the Internet?
Choose from a list of Internet service providers (ISPs)
Set up my connection manually
For a dial-up connection, you will need your account name, password, and a phone number for your ISP. For a broadband account, you won't need a phone number.
Use the CD I got from an ISP
< Back
Next >
Cancel

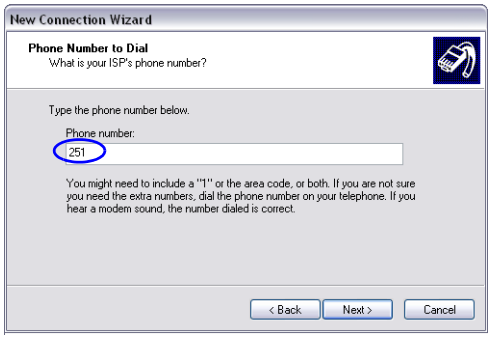
5 In the *Internet Connection* box, select **Connect using dial-up modem** and click the **Next** button.

New Connection Wizard
Internet Connection
How do you want to connect to the Internet?
Connect using a dial-up modem
This type of connection uses a modem and a regular or ISDN phone line.
Connect using a broadband connection that requires a user name and password
This is a high-speed connection using either a DSL or cable modem. Your ISP may refer to this type of connection as PPPoE.
Connect using a broadband connection that is always on
This is a high-speed connection using either a cable modem, DSL or LAN connection. It is always active, and doesn't require you to sign in.
< Back
Next >
Cancel

6 In the *Connection Name* box, enter a **Name** for the modem connection (e.g. *Modem Connection*) and click the **Next** button.


New Connection Wizard
Connection Name
What is the name of the service that provides your Internet connection?
Type the name of your ISP in the following box.
ISP Name
Modem Connection
The name you type here will be the name of the connection you are creating.
< Back
Next >
Cancel

7 In the *Phone Number to Dial* box, enter the **Phone Number** for the modem and click the **Next** button.


New Connection Wizard
Phone Number to Dial
What is your ISP's phone number?
Type the phone number below.
Phone number:
251
You might need to include a "1" or the area code, or both. If you are not sure you need the extra numbers, dial the phone number on your telephone. If you hear a modem sound, the number dialed is correct.
< Back
Next >
Cancel

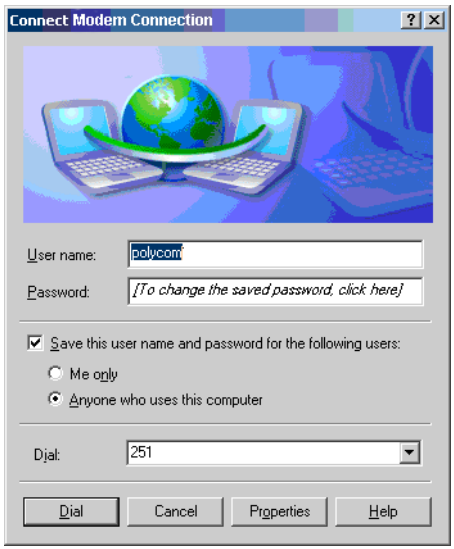
8 In the *Connection Availability* box, select **Anyone's use** and click the **Next** button.

New Connection Wizard
Connection Availability
You can make the new connection available to any user or only to yourself.
A connection that is created for your use only is saved in your user account and is not available unless you are logged on.
Create this connection for:
Anyone's use
My use only
< Back
Next >
Cancel

9 In the *Internet Account Information* box, complete the *Username*, *Password* and *Confirm Password* fields and click the **Next** button.

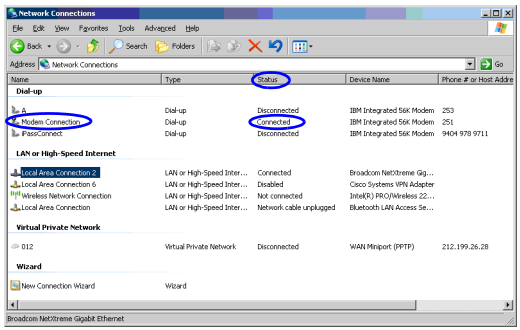
New Connection Wizard
Internet Account Information
You will need an account name and password to sign in to your Internet account.
Type an ISP account name and password, then write down this information and store it in a safe place. (If you have forgotten an existing account name or password, contact your ISP.)
User name:
polycom
Password:
Confirm password:
Use this account name and password when anyone connects to the Internet from this computer
Make this the default Internet connection
< Back
Next >
Cancel

10 The *Connection* applet is displayed with the field values filled in as specified by the *New Connection Wizard*.



11 Click the **Dial** button to establish a connection to *LAN 3 Port* via the modem.

The *Windows – Network Connections* applet displays *Connected* status for the new connection.



Procedure 4: Connect to the RMX

To Connect using the RMX Manager:

To use the browser:

- In the browser's command line, enter **`http://<MCU Control Unit IP Address>/RmxManager.html`** and press **Enter**.

To use the Windows Start menu:

1 Click **Start.**

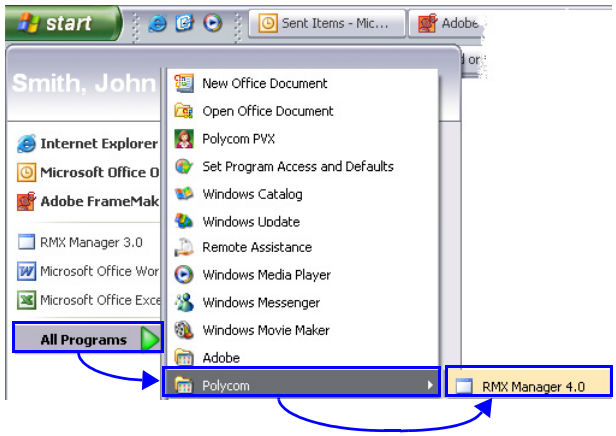
a If the *RMX Manager* appears in the recently used programs list, click **RMX Manager** in the list to start the application.

or

b Click **All Programs.**

The *All Programs* list is displayed.

a Select **Polycom** and then select **RMX Manager**.



The *RMX Manager – Welcome* screen is displayed.

Appendix H

Setting the RMX for Integration Into Microsoft OCS Environment

Point-to-point and multipoint audio and video meetings can be initiated from Office Communicator, Windows Messenger and Polycom video endpoints (HDX and VSX) when the environment components are installed and configured.

Multipoint calls are enabled when the RMX is installed in this environment and the following configuration procedures have been completed:

1. Set the Static Route & Trusted Host for RMX in the OCS.
2. **Optional.** Creating the security (TLS) certificate in the OCS and exporting the certificate to the RMX workstation. The certificate files can also be obtained from a Certificate Authority.
3. **Optional if Load Balancer Server is present.** Set the Static Route & Trusted Host for RMX in the Load Balancer server.
4. Modify the Management Network Service to include the DNS server and set the Transport Type to TLS.
5. Define a SIP Network Service in the RMX and install the TLS certificate.
6. Modify and add the required system flags in the RMX System Configuration.
7. **Optional.** Defining additional Entry Queues and Meeting Rooms in the RMX environment. For details see the *RMX Administrator's Guide*.

For a detailed description of the configuration of the Polycom conferencing entities for the integration in Microsoft Office Communications Server 2007 see *Polycom® HDX and RMX™ Systems Integration with Microsoft Office Communications Server 2007 Deployment Guide.*

Configuring the OCS for RMX 2000

Setting the Trusted Host and Static Route for RMX in the OCS

To be able to work with the OCS, the RMX unit must be configured as a Trusted Host in the OCS. This is done by defining the IP address of the signaling host of each RMX unit as Trusted Host.

Meeting Rooms are usually not registered to the OCS, and Static Routes are used instead. Setting Static Routes in the OCS enables SIP entities / UAs to connect to conferences without explicit registration of conferences with the OCS.

Routing is performed by the OCS based on the comparison between the received URI and the provisioned static route pattern. If a match is found, the request is forwarded to the next hop according to the defined hop's address.

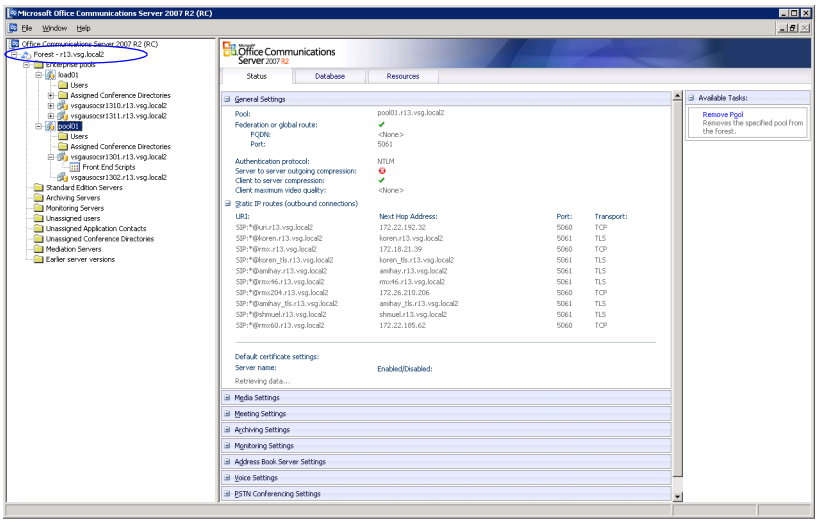
This is the recommended working method. It alleviates the need to create a user account in the OCS for each Meeting Room and Entry Queue. This also allows users to join ongoing conferences hosted on the MCU without registering all these conferences with OCS.

Entry Queues can also be for Ad-hoc conferencing enabling Office Communicator clients to dial to the Entry Queue and create a new ongoing conference using DTMF codes to enter the target conference ID. In such a case, other OC users will have to use that ID to join the newly created conference.

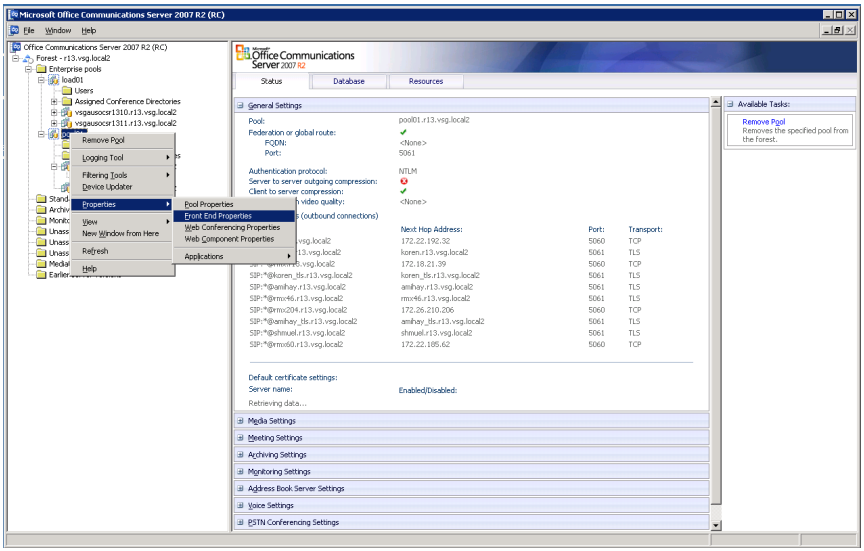
To set the RMX as trusted and define Static Routes in OCS:

1 Open the OCS Management application.

2 Expand the *Enterprise Pools* list.

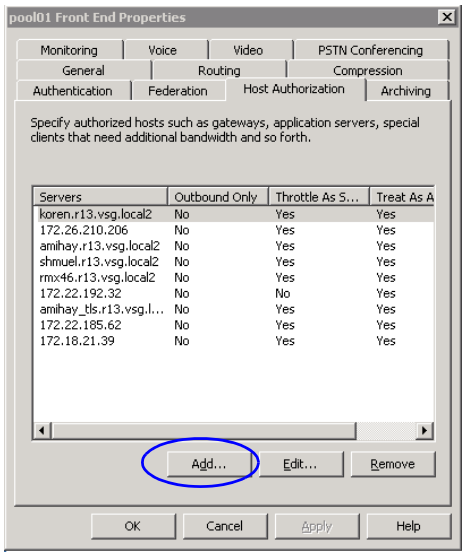


3 Right-click the *server pool* icon, click **Properties > Front End Properties**.

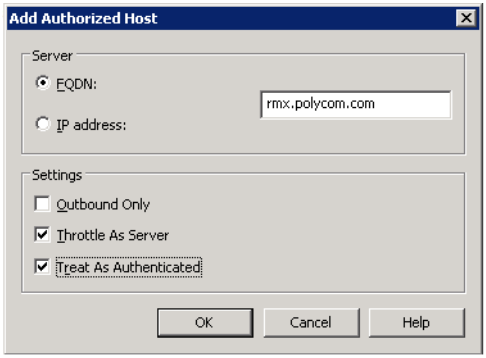


The *Pool Front End Properties* dialog box opens.

4 Click the **Host Authorization** tab.



5 Click the **Add** button to add the RMX as trusted host.
The *Add Authorized Host* dialog box opens.



6 In the *Add Authorized Host* dialog box, enter the RMX *FQDN* name as defined in the DNS and will be used in the Static Routes definition.

7 In the *Settings* section, select the **Throttle as Server** and **Treat As Authenticated** check boxes.

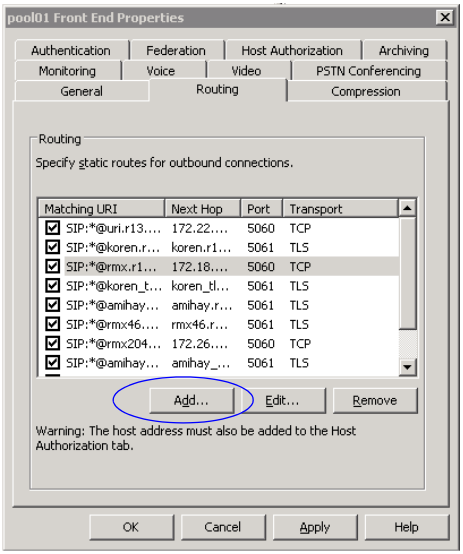
8 Click **OK**.
The defined RMX appears in the trusted servers list in the server *Front End Properties – Host Authorization* dialog box.

If routing between the RMX and the OCS using Static Routes is required, do not close this dialog box, and continue with the following procedure. If you do not want to define Static Routes, click OK to close this dialog box.

To add RMX to the Routing Roles:

9 In the *Front End Properties* dialog box, click the **Routing** tab.

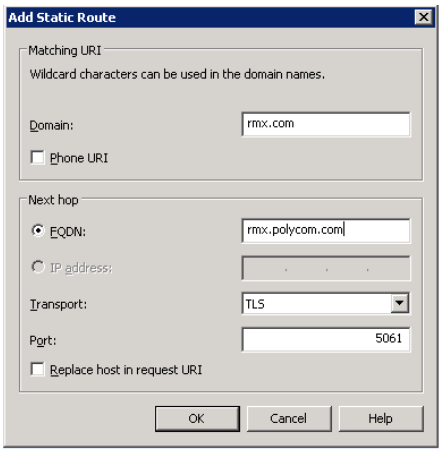
10 Click the **Add** button.



The *Add Static Routes* dialog box opens.

11 In the *Matching URI* section, enter the *Domain* name for the RMX. Any domain name can be used.

12 In the *Next hop* section enter the RMX *FQDN* name as defined in the DNS and is used in the *Host Authorization* definition.



13 In the *Transport* field, select **TLS** to enable the dial-out from conferences to SIP endpoints.

14 Click **OK**.
The new Route is added to the list of routes in the *Front End Properties – Routes* dialog box.

15 Click **OK**.

Creating the Security (TLS) Certificate in the OCS and Exporting the Certificate to the RMX Workstation

To work in Microsoft R1 and R2 environment or when encryption of SIP signaling is used, the SIP server and the RMX *Transport Type* must be set to **TLS** and a certificate must be created and sent to the RMX.

If a Load Balancer is used in Microsoft R1environment, the transport type may be set to TCP or TLS.

In this scenario, a video conference is scheduled on a Polycom MCU and it includes predefined participants; Office Communicator and other SIP and non-SIP users. At the scheduled time the conference is activated and the MCU automatically dials out to the predefined participants and connects them to the conference.

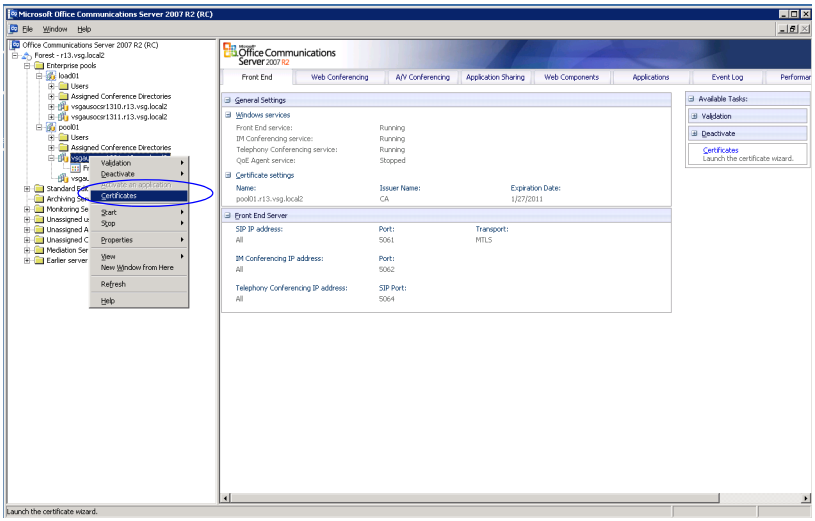
To enable the TLS transport, certificate files *rootCA.pem*, *pkey.pem* and *cert.pem* must be sent to the RMX unit. These files can be created and sent to the RMX in two methods:

- The files *rootCA.pem*, *pkey.pem* and *cert.pem* are provided by a Certificate Authority and are sent independently or together with a password file to the RMX. This is the recommended method.
- Alternatively, the TLS certificate files are created internally in the OCS and exported to the RMX workstation from where the files can be downloaded to the RMX. If the certificate is created internally by the OCS, one **.pfx* file is created. In addition, a text file containing the password that was used during the creation of the **.pfx* file is manually created. Both files can then be sent from the RMX workstation to the RMX unit. When the files are sent to the RMX, the **.pfx* file is converted into three certificate files: *rootCA.pem*, *pkey.pem* and *cert.pem*.
  Sometimes, the system fails to read the *.pfx file and the conversion process fails. Resending *.pfx file again and then resetting the system may resolve the problem.

To create the TLS certificate in the OCS:

1 In the OCS *Enterprise Pools* tree, expand the Pools list and the *server pool* list.

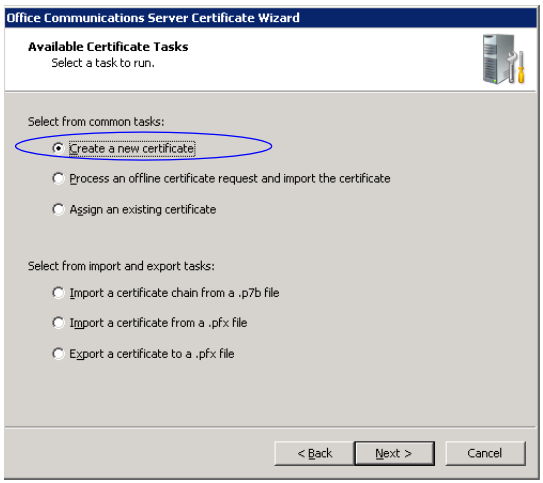
2 Right-click the pool *Front End* entity, and click **Certificate**.



The *Office Communicator Server Wizard Welcome* window is displayed.

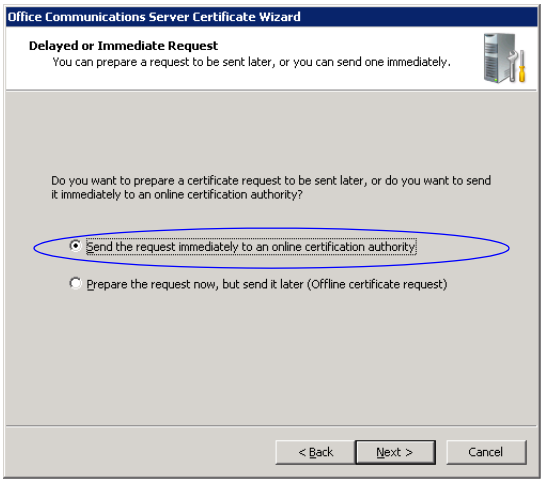
3 Click **Next**.
The *Available Certificate Tasks* window appears.

4 Select **Create a New Certificate** and click **Next**.


Office Communications Server Certificate Wizard
Available Certificate Tasks
Select a task to run.
Select from common tasks:
Create a new certificate
Process an offline certificate request and import the certificate
Assign an existing certificate
Select from import and export tasks:
Import a certificate chain from a .p7b file
Import a certificate from a .pfx file
Export a certificate to a .pfx file
< Back
Next >
Cancel

The *Delayed or Immediate Request* window appears.

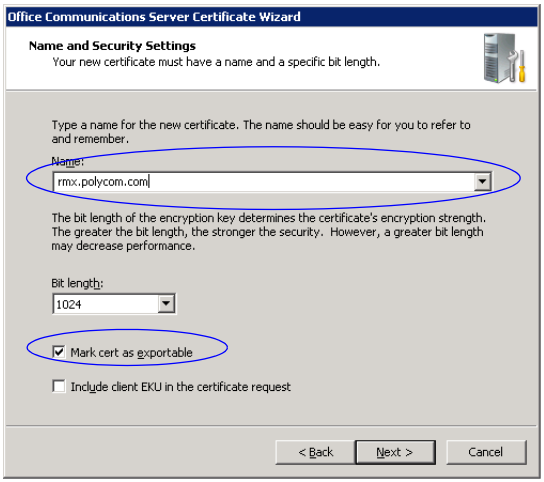
5 Select **Send the Request immediately to an online certificate authority** and click **Next**.


Office Communications Server Certificate Wizard
Delayed or Immediate Request
You can prepare a request to be sent later, or you can send one immediately.
Do you want to prepare a certificate request to be sent later, or do you want to send it immediately to an online certification authority?
Send the request immediately to an online certification authority
Prepare the request now, but send it later (Offline certificate request)
< Back
Next >
Cancel

The *Name and Security Settings* window appears.

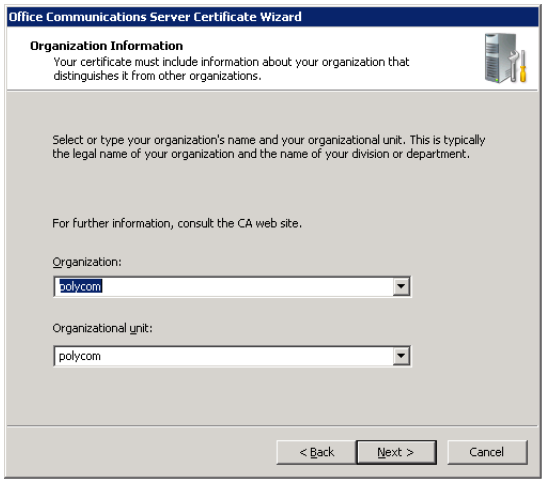
6 In the *Name* field, select the RMX name you entered in the *FQDN* field when defining the trusted host or as defined in the DNS server.

7 Select the **Mark cert as exportable** check box.


Office Communications Server Certificate Wizard
Name and Security Settings
Your new certificate must have a name and a specific bit length.
Type a name for the new certificate. The name should be easy for you to refer to and remember.
Name:
rmx.polycom.com
The bit length of the encryption key determines the certificate's encryption strength. The greater the bit length, the stronger the security. However, a greater bit length may decrease performance.
Bit length:
1024
Mark cert as exportable
Include client EKU in the certificate request
< Back
Next >
Cancel

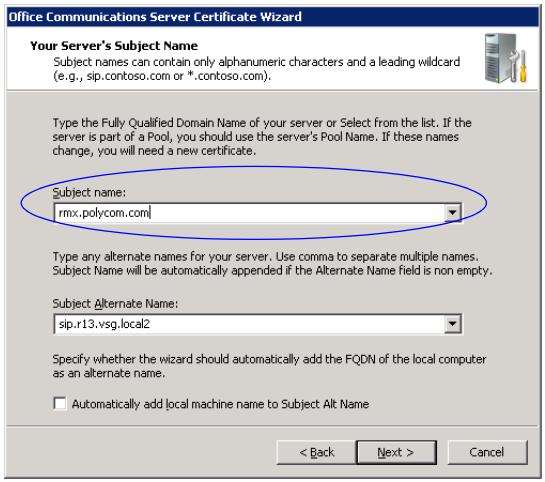
8 Click **Next**.
The *Organization Information* window appears.

9 Enter the name of the *Organization* and the *Organization Unit* and click **Next**.


Office Communications Server Certificate Wizard
Organization Information
Your certificate must include information about your organization that distinguishes it from other organizations.
Select or type your organization's name and your organizational unit. This is typically the legal name of your organization and the name of your division or department.
For further information, consult the CA web site.
Organization:
polycom
Organizational unit:
polycom
< Back
Next >
Cancel

Your Server's Subject Name window appears.

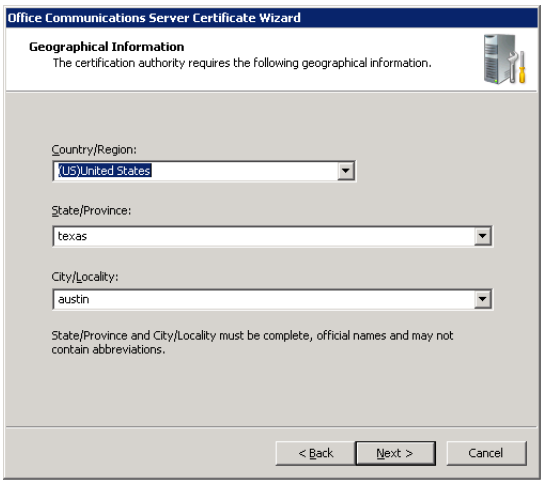
10 In the *Subject name* field, select the *FQDN* name of the RMX from the list or enter its name.
Keep the default selection in the *Subject alternate name* field and click **Next**.


Office Communications Server Certificate Wizard
Your Server's Subject Name
Subject names can contain only alphanumeric characters and a leading wildcard (e.g., sip.contoso.com or *.contoso.com).
Type the Fully Qualified Domain Name of your server or Select from the list. If the server is part of a Pool, you should use the server's Pool Name. If these names change, you will need a new certificate.
Subject name:
rmx.polycom.com
Type any alternate names for your server. Use comma to separate multiple names. Subject Name will be automatically appended if the Alternate Name field is non empty.
Subject Alternate Name:
sip.r13.vsg.local2
Specify whether the wizard should automatically add the FQDN of the local computer as an alternate name.
Automatically add local machine name to Subject Alt Name
< Back
Next >
Cancel

11 If an error message is displayed, click **Yes** to continue.

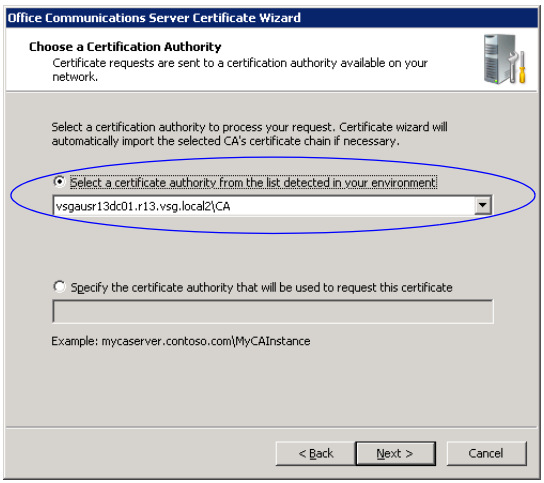
The *Geographical Information* window appears.

12 Enter the geographical information as required and click **Next**.



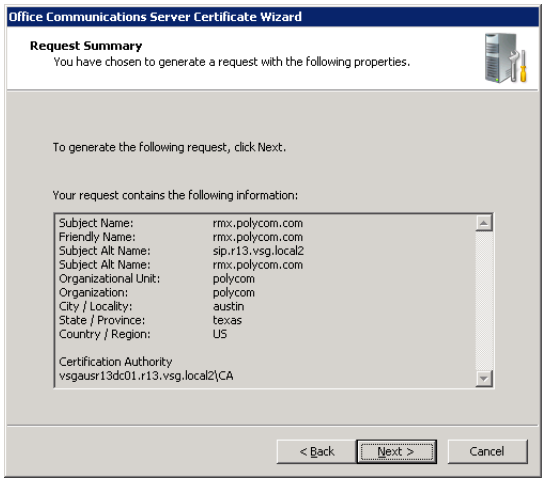
The *Choose a Certification Authority* window appears.

13 Ensure that the **Select a certificate authority from the list detected in your environment** option is selected and that the local OCS front end entity is selected.


Office Communications Server Certificate Wizard
Choose a Certification Authority
Certificate requests are sent to a certification authority available on your network.
Select a certification authority to process your request. Certificate wizard will automatically import the selected CA's certificate chain if necessary.
Select a certificate authority from the list detected in your environment
vsgausr13dc01.r13.vsg.local2\CA
Specify the certificate authority that will be used to request this certificate
Example: mycaserver.contoso.com\MyCAInstance
< Back
Next >
Cancel

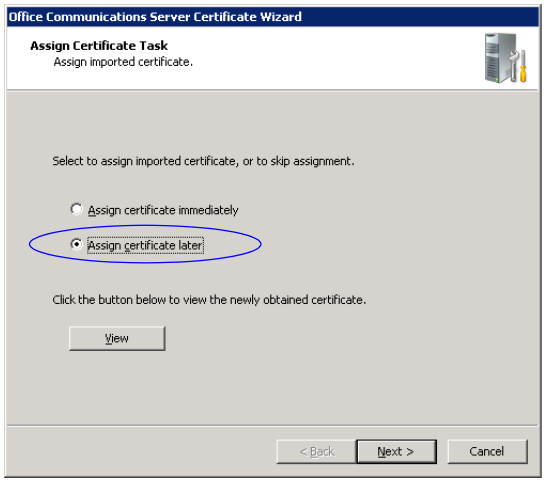
14 Click **Next**.
The *Request Summary* window appears.

15 Click **Next** to confirm the listed parameters and create the requested certificate.



The *Assign Certificate Task* window appears.

16 Select **Assign certificate later** and click **Next** (MS R2).
Select **Assign certificate later** and click **Finish** (MS R1).


Office Communications Server Certificate Wizard
Assign Certificate Task
Assign imported certificate.
Select to assign imported certificate, or to skip assignment.
Assign certificate immediately
Assign certificate later
Click the button below to view the newly obtained certificate.
View
< Back
Next >
Cancel

The *Certificate Wizard Completed* window appears (MS R2).

17 Click **Finish** (MS R2).

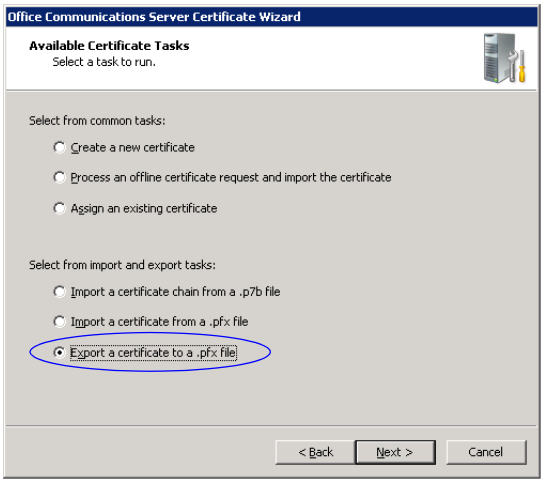
Retrieving the Certificate from the OCS to the RMX Workstation

1 In the OCS *Enterprise Pools* tree, expand the *Pools* list and the *Server Pool* list.

2 Right-click the *pool Front End* entity, and select **Certificate**.

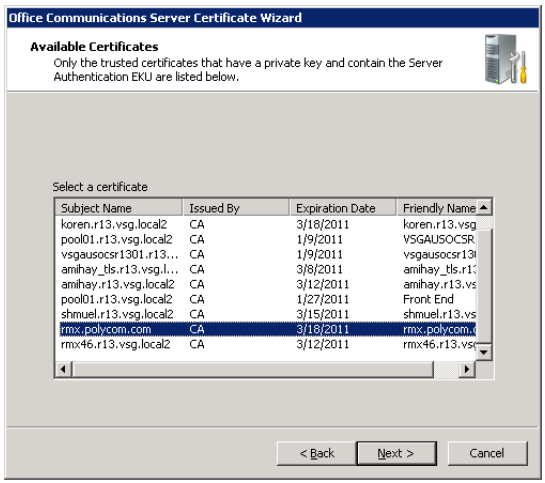
The *Available Certificate Tasks* window appears.

3 Select **Export a certificate to a *.pfx file** and click **Next**.


Office Communications Server Certificate Wizard
Available Certificate Tasks
Select a task to run.
Select from common tasks:
Create a new certificate
Process an offline certificate request and import the certificate
Assign an existing certificate
Select from import and export tasks:
Import a certificate chain from a .p7b file
Import a certificate from a .pfx file
Export a certificate to a .pfx file
< Back
Next >
Cancel

The *Available Certificates* window appears.

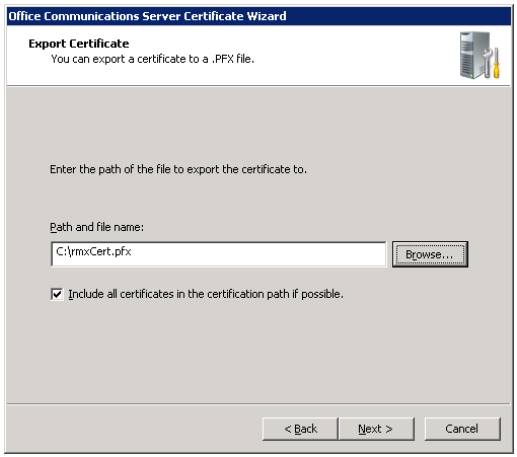
4 Select the certificate *Subject Name* of the RMX and click **Next.**


Office Communications Server Certificate Wizard
Available Certificates
Only the trusted certificates that have a private key and contain the Server Authentication EKU are listed below.
Select a certificate
Subject Name
Issued By
Expiration Date
Friendly Name
koren.r13.vsg.local2 CA 3/18/2011 koren.r13.vsg
pool01.r13.vsg.local2 CA 1/9/2011 VSGAUSOCSR
vsgausocsr1301.r13... CA 1/9/2011 vsgausocsr13
amihay_tls.r13.vsg.l... CA 3/8/2011 amihay_tls.r1
amihay.r13.vsg.local2 CA 3/12/2011 amihay.r13.vs
pool01.r13.vsg.local2 CA 1/27/2011 Front End
shmuel.r13.vsg.local2 CA 3/15/2011 shmuel.r13.vs
rmx.polycom.com CA 3/18/2011 rmx.polycom.
rmx46.r13.vsg.local2 CA 3/12/2011 rmx46.r13.vs
< Back
Next >
Cancel

The *Export Certificate* window appears.

5 Enter the path and file name of the certificate file to be exported or click the **Browse** button to select the path from the list.

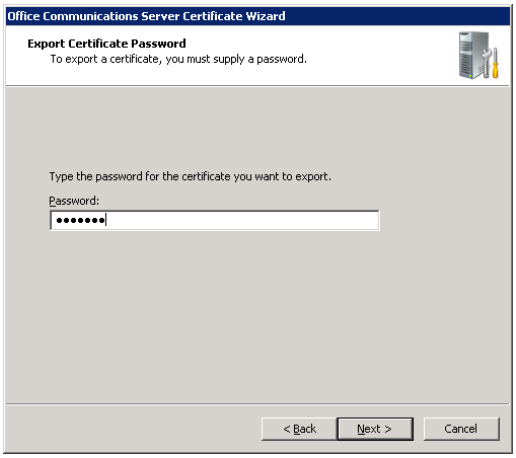
The new file type must be ***.pfx** and its name must include the **.pfx** extension.


Office Communications Server Certificate Wizard
Export Certificate
You can export a certificate to a .PFX file.
Enter the path of the file to export the certificate to.
Path and file name:
C:\rmxCert.pfx
Browse...
Include all certificates in the certification path if possible.
< Back
Next >
Cancel

6 Select the **Include all certificates in the certification path if possible** check box and then click **Next**.

The *Export Certificate Password* window appears.

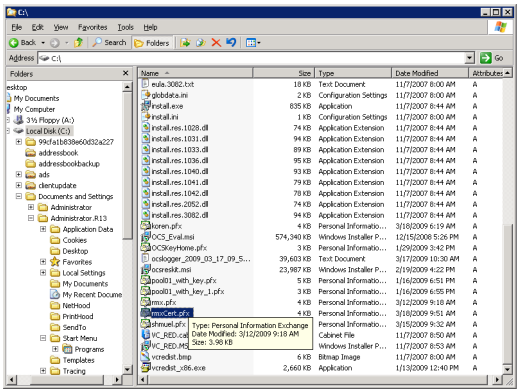
7 If required, enter any password. For example, *Polycom*.
Write down this password as you will have to manually create a password file in which this password will appear.


Office Communications Server Certificate Wizard
Export Certificate Password
To export a certificate, you must supply a password.
Type the password for the certificate you want to export.
Password:
< Back
Next >
Cancel

Click **Next**.
The *Certificate Wizard Completed* window appears.

8 Click **Finish**.

The created **.pfx* file is added in the selected folder.



Optional. Creating the Certificate Password File (certPassword.txt)

If you have used a password when creating the certificate file (*.*pfx*), you must create a **certPassword.txt** file. This file will be sent to the RMX together with the *.*pfx* file.

To create the certPassword.txt file:

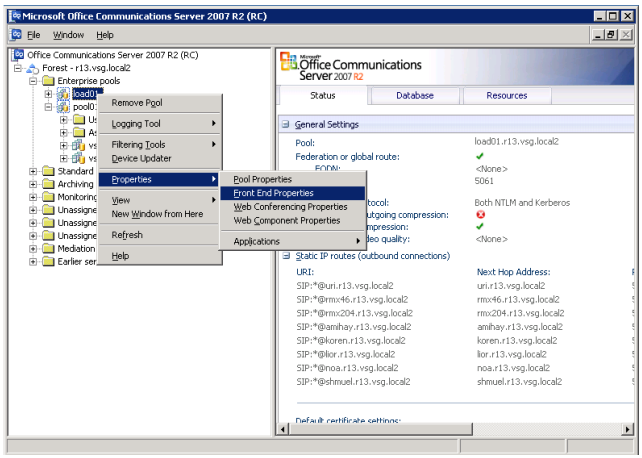
1. Using a text editor application, create a new file.
2. Type the password as you have entered when creating the certificate file. For example, enter *Polycom*.
3. Save the file naming it **certPassword.txt** (file name must be exactly as show, the RMX is case sensitive).

Optional. Setting the Static Route & Trusted Host for RMX in the Load Balancer Server

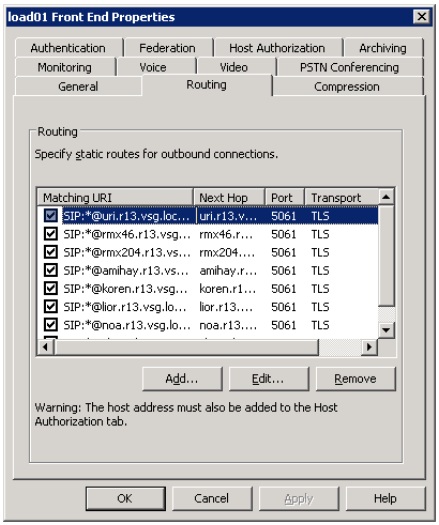
If your network includes a Load Balancer server, the RMX unit must be configured as a trusted host in the Load Balancer server in the same way it is configured in the OCS. In addition, Static Routes must also be defined in the Load Balancer server in the same way it is configured in the OCS. This configuration procedure is done in addition to the configuration in the OCS.

To set the RMX as trusted and define Static routes in the Load Balancer Server:

1. Open the OCS Management application.
2. Expand the *Enterprise Pools* list.
3. Right-click the *Load* icon, click **Properties > Front End Properties**.



The *Load Front End Properties* dialog box opens.



The definition procedure is the same as for setting the RMX as trusted and define Static routes in the OCS. For details, see "*Setting the Trusted Host and Static Route for RMX in the OCS*" on page **2**.

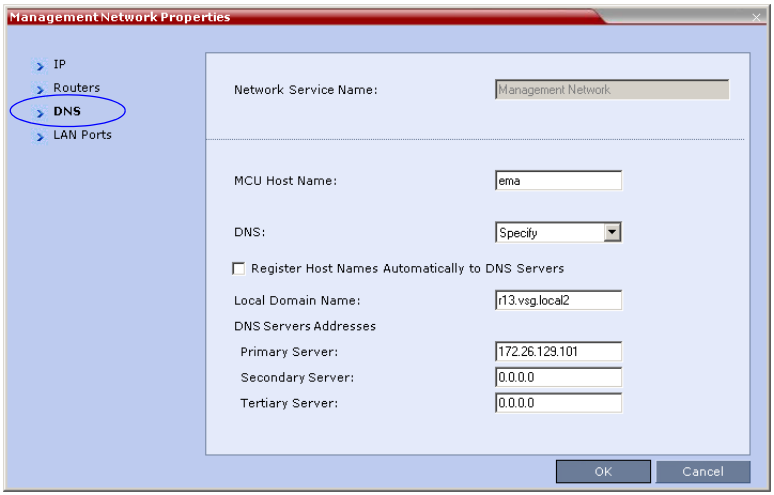
Configuring the RMX 2000 for Microsoft OCS 2007 Integration

Modify the RMX Management Network Service to Include the DNS Server

The Management Network that is defined during first entry setup does not include the definition of the DNS which is mandatory in Microsoft environment and has to be modified.

To add the definition of the DNS to the Management Network in the RMX:

1. Using the Web browser, connect to the RMX.
2. In the *RMX Management* pane, expand the **Rarely Used** list and click **IP Network Services** ( ).
3. In the *IP Network Services* pane, double-click the **Management Service** .
   The *Management Network Properties - IP* dialog box opens.
4. Click the **DNS** tab.



5. In the *DNS* field, select **Specify** to define the DNS parameters.

6 View or modify the following fields:

Table 1 Management Network Properties – DNS Parameters

Field	Description
MCU Host Name	Enter the name of the MCU on the network. This name must be identical to the FQDN name defined for the RMX in the OCS and DNS. Default name is RMX.
Shelf Management Host Name	Displays the name of the entity that manages the RMX hardware. The name is derived from the MCU host name. Default is RMX_SHM.
DNS	Select: • **Off** – if DNS servers are not used in the network. • **Specify** – to enter the IP addresses of the DNS servers. **Note:** The IP address fields are enabled only if **Specify** is selected.
Register Host Names Automatically to DNS Servers	Select this option to automatically register the MCU Signaling Host and Shelf Management with the DNS server.
Local Domain Name	Enter the name of the domain where the MCU is installed as defined in the OCS.
DNS Servers Addresses:	
Primary Server	The static IP addresses of the DNS servers (the same servers defined in the OCS). A maximum of three servers can be defined.
Secondary Server	
Tertiary Server	

7 Click **OK**.

Defining a SIP Network Service in the RMX

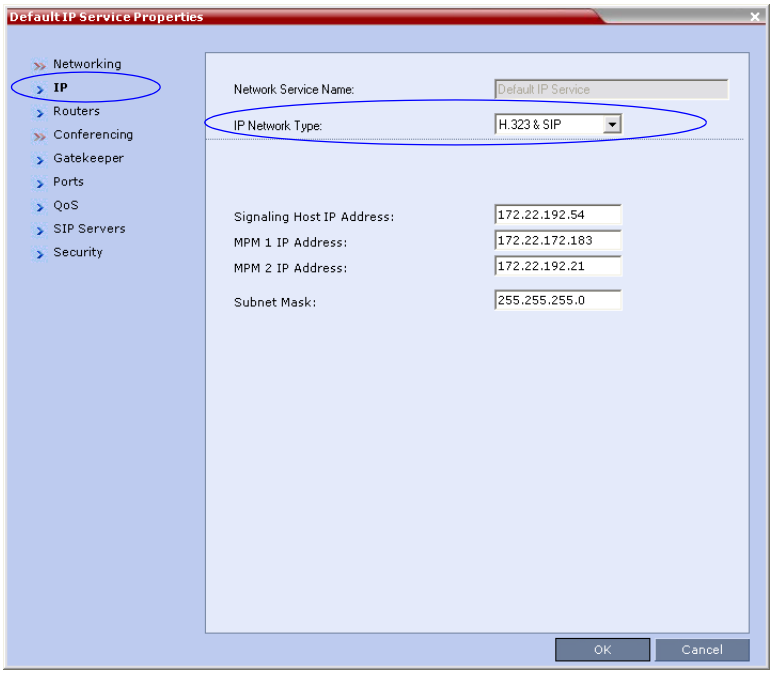
Your Polycom RMX 2000 system should be installed according to standard installation procedures. See the *Polycom RMX 2000 Getting Started Guide*, which describes how to set up and configure the MCU.

When configuring the Default IP Network Service on first entry, or when modifying the properties of the existing Default IP Network Service, the SIP environment parameters must be set as described in "*Defining a SIP Network Service in the RMX*" on page **21**.

To configure the RMX 2000 IP Network Service:

1. Using the Web browser, connect to the RMX.
2. In the *RMX Management* pane, expand the **Rarely Used** list and click **IP Network Services** ( ).
3. In the *IP Network Services* pane, double-click the **Default IP Service** (, , or) entry.

   The *Default IP Service - Networking IP* dialog box opens.

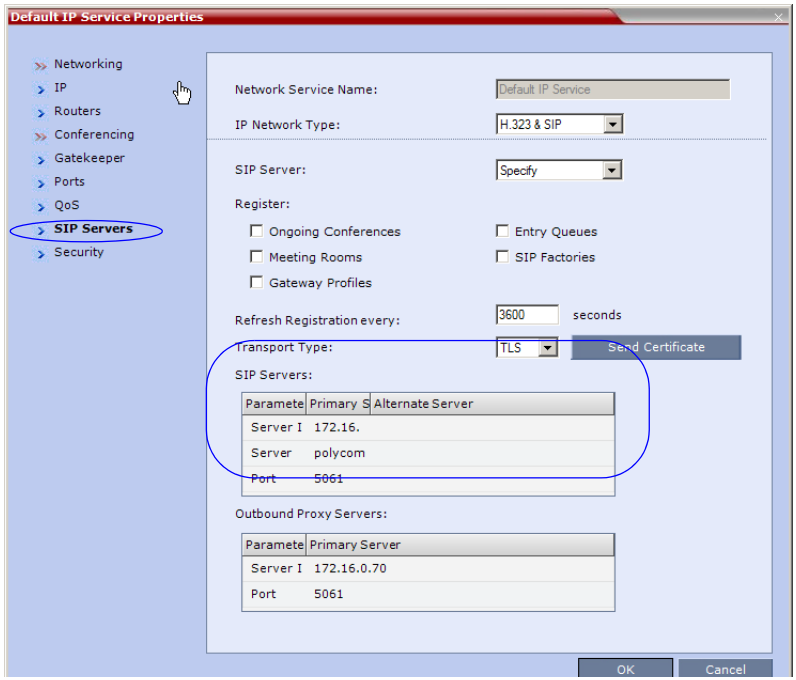


4 Make sure the *IP Network Type* is set to **H.323 & SIP** even though SIP will be the only call setup used with Office Communications Server 2007.

5 Make sure that the correct parameters are defined for the *Signaling Host IP Address*, *MPM 1 IP Address*, *MPM 2 IP Address* (if necessary), and *Subnet Mask*.

Make sure that the IP address of the RMX Signaling Host is the same one defined as a trusted host in Office Communications Server 2007.

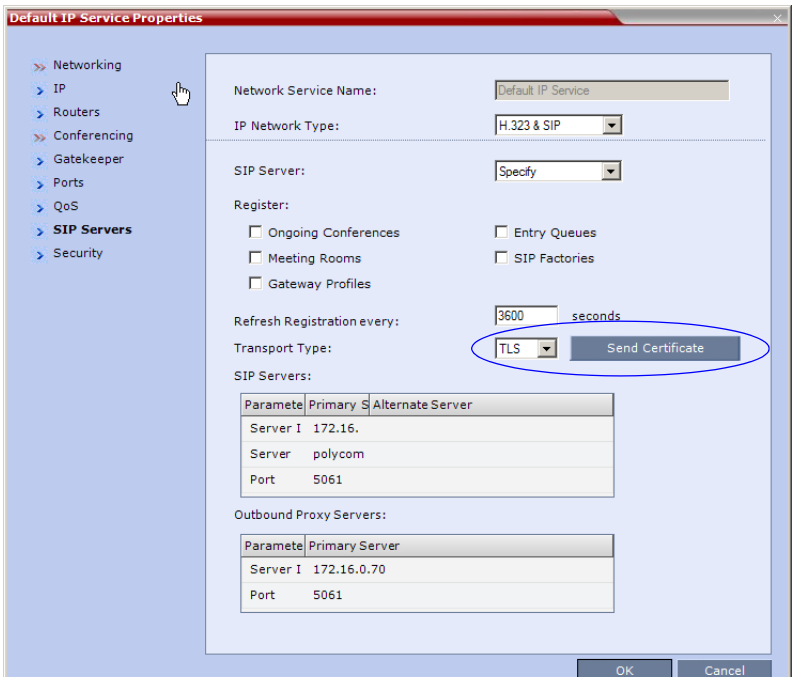
6 Click the **SIP Servers** tab.



7 Make sure the IP address of the Office Communications Server 2007 is specified and the *Server Domain Name* is the same as defined in the OCS and in the *Management Network* for the DNS.

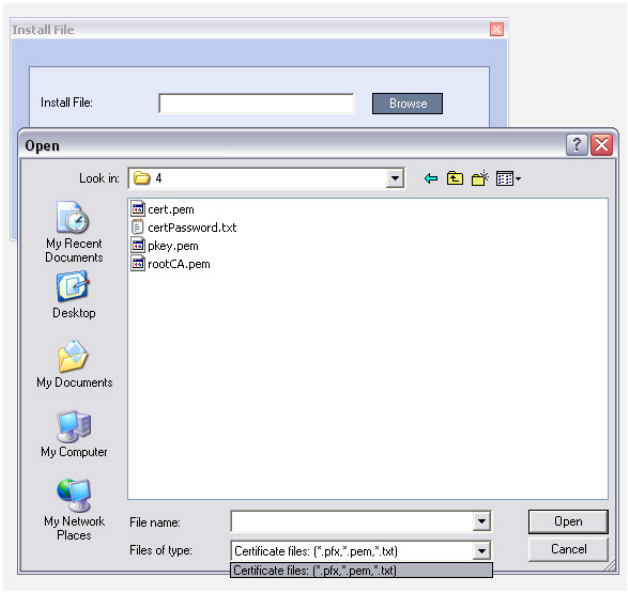
8 Change the *Transport Type* to **TLS**.
The *Send Certificate* button is enabled.

9 Click the **Send Certificate** button.



The *Install File* dialog box opens.

10 Click the **Browse** button.
The *Open* dialog box appears, letting you select the certificate file(s) to send to the MCU.



Depending on the method used when the certificate file(s) were created, send the certificate file(s) to the RMX according to the contents of the file set that was created:

— The certificate files *rootCA.pem, pkey.pem*, *cert.pem* and a *certPassword.txt*. The files were created by a Certificate Authority and are sent as is to the RMX together with the required password contained in the *certPassword.txt* file.
This is the recommended method.

— The files *rootCA.pem, pkey.pem* and *cert.pem*. The certificate files were created by a Certificate Authority and are sent as is to the RMX.

— A **.pfx* file and a *certPassword.txt* file. The file c*ertPassword.txt* is manually created if the **.pfx* file was created by the OCS using a password. The **.pfx* file will be converted internally by the RMX using the password included in the certPassword.txt into three certificate files named *rootCA.pem*, *pkey.pem* and *cert.pem*.

— A **.pfx* file if the certificate file was created in the OCS without using a password. The **.pfx* file will be converted internally by the RMX into three certificate files named r*ootCA.pem*, *pkey.pem* and *cert.pem*.

11 In the file browser, select all files to be sent in one operation according to the contents of the set:

— One ***.pfx** file, or

— Two files: one ***.pfx** file and **certPassword.txt**, or

— Three files: **rootCA.pem, pkey.pem** and **cert.pem**, or

— Four files: **rootCA.pem, pkey.pem, cert.pem** and **certPassword.txt**

12 Click **Open**.
The selected file(s) appear in the *Install Files* path.

13 Click **Install**.
The files are sent to the RMX and the *Install File* dialog box closes.

14 In the *Default IP Service - Networking IP* dialog box, click **OK**.

15 In the *Reset Confirmation* dialog box, click **No** to modify the required system flags before resetting the MCU, or click **Yes** if the flag was already set**.**

Reset can be performed after setting the system flags (for example, setting the MS_ENVIRONMENT flag). After system reset the RMX can register to the OCS server and make SIP calls.
Sometimes the system fails to read the *.pfx file and the conversion process fails, which is indicated by the active alarm "SIP TLS: Registration server not responding" and/or "SIP TLS: Registration handshake failure". Sending *.pfx file again, as described in this procedure and then resetting the system may resolve the problem.

Polycom RMX System Flag Configuration

The RMX can be installed in Microsoft R1 or R2 environments. To adjust the RMX behavior to the Microsoft environment in each release, system flags must be set.

To configure the system flags on the Polycom RMX 2000 system:

1 On the *RMX* menu, click **Setup > System Configuration**.
The *System Flags - MCMS_PARAMETERS_USER* dialog box opens.

2 Scroll to the flag **MS_ENVIRONMENT** and click it. The *Edit Flag* dialog box is displayed.

3 In the *Value* field, enter **YES** to set the RMX SIP environment to Microsoft solution.

RMX set to MS_ENVIRONMENT=YES supports SIP over TLS only and not over TCP.

4 Click **OK** to complete the flag definition.

5 When prompted, click **Yes** to reset the MCU and implement the changes to the system configuration. After system reset the RMX can register to the OCS server and make SIP calls.

Sometimes the system fails to read the *.pfx file and the conversion process fails, which is indicated by the active alarm “SIP TLS: Registration server not responding” and/or “SIP TLS: Registration handshake failure”. Sending *.pfx file again, as described in this procedure and then resetting the system may resolve the problem.

In some configurations, the following flags may require modifications when **MS_ENVIRONMENT** flag is set to YES:

Table H-1 *Additional Microsoft Environment Flags in the RMX MCMS_PARAMETERS_USER Tab*

Flag Name	Value and Description
SIP_FREE_VIDEO_RESOURCES	Default value in Microsoft environment: **NO.** When set to NO, video resources that were allocated to participants remain allocated to the participants as long as they are connected to the conference even if the call was changed to audio only. The system does not allocate the resources to other participants ensuring that the participant have the appropriate resources in case they want to return to the video call.

Table H-1 *Additional Microsoft Environment Flags in the RMX MCMS_PARAMETERS_USER Tab*

Flag Name	Value and Description
SIP_FREE_VIDEO_RESOURCES (continued)	The system allocates the resources according to the participant's endpoint capabilities, with a minimum of one CIF video resource. Enter **YES** to enable the system to free the video resources for allocation to other conference participants. The call becomes an audio only call and video resources are not guaranteed to participants if they want to add video again.
SIP_FAST_UPDATE_INTERVAL_ ENV	Default setting is **0** to prevent the RMX from automatically sending an Intra request to all SIP endpoints. Enter **n** (where n is any number of seconds other than 0) to let the RMX automatically send an Intra request to all SIP endpoints every n seconds. It is recommended to set the flag to 0 and modify the frequency in which the request is sent at the endpoint level (as defined in the next flag).
SIP_FAST_UPDATE_INTERVAL_ EP	Default setting is 6 to let the RMX automatically send an Intra request to Microsoft OC endpoints only, every 6 seconds. Enter any other number of seconds to change the frequency in which the RMX send the Intra request to Microsoft OC endpoints only. Enter 0 to disable this behavior at the endpoint level (not recommended).

Dialing to an Entry Queue, Meeting Room or Conference

The preferred dialing mode to the conferencing entities such as Meeting Rooms, conferences and Entry Queues is direct dial in using the domain name defined in the OCS Static Routes. This eliminates the need to register the conferencing entities with the SIP server and to define a separate user for each conferencing entity in the Active Directory.

In such a case, after the first dial in, the conferencing entity will appear in the OC client directory for future use.

To dial in directly to a conference or Entry Queue:

Enter the conferencing entity SIP URI in the format:
conferencing entity routing name@domain name

The domain name is identical to the domain name defined in the OCS Static Routes.

For example, if the domain name defined in the OCS static routes is lcs2007.polycom.com and the Routing Name of the Meeting Room is 4567, the participant enters 4567@lcs2007.polycom.com.

Another dialing method is to register the Entry Queues with the SIP Server and create a user for each Entry Queue in the Active Directory. In such a case, OC clients can select the Entry Queue from the Contacts list and dial to the Entry Queue.

Active Alarms and Troubleshooting

Active Alarms

The following active alarms may be displayed in the RMX *System Alerts* pane when the RMX is configured for integration in the OCS environment:

***Table H-2** New Active Alarms*

Alarm Code	Alarm Description
SIP TLS: Failed to load or verify certificate files	This alarm indicates that the certificate files required for SIP TLS could not be loaded to the RMX. Possible causes are: • Incorrect certificate file name. Only files with the following names can be loaded to the system: rootCA.pem, pkey.pem, cert.pem and certPassword.txt • Wrong certificate file type. Only files of the following types can be loaded to the system: rootCA.pem, pkey.pem and cert.pem and certPassword.txt • The contents of the certificate file does not match the system parameters
SIP TLS: Registration transport error	This alarm indicates that the communication with the SIP server cannot be established. Possible causes are: • Incorrect IP address of the SIP server • The SIP server listening port is other than the one defined in the system • The OCS services are stopped **Note:** Sometimes this alarm may be activated without real cause. Resetting the MCU may clear the alarm.

Table H-2 New Active Alarms (Continued)

Alarm Code	Alarm Description
SIP TLS: Registration handshake failure	This alarm indicates a mismatch between the security protocols of the OCS and the RMX, preventing the Registration of the RMX to the OCS.
SIP TLS: Registration server not responding	This alarm is displayed when the RMX does not receive a response from the OCS to the registration request in the expected time frame. Possible causes are: • The RMX FQDN name is not defined in the OCS pool, or is defined incorrectly. • The time frame for the expected response was too short and it will be updated with the next data refresh. The alarm may be cleared automatically the next time the data is refreshed. Alternatively, the OCS Pool Service can be stopped and restarted to refresh the data. • The RMX FQDN name is not defined in the DNS server. Ping the DNS using the RMX FQDN name to ensure that the RMX is correctly registered to the DNS.
SIP TLS: Certificate has expired	The current TLS certificate files have expired and must be replaced with new files.
SIP TLS: Certificate is about to expire	The current TLS certificate files will expire shortly and will have to be replaced to ensure the communication with the OCS.
SIP TLS: Certificate subject name is not valid or DNS failed to resolve this name	This alarm is displayed if the name of the RMX in the certificate file is different from the FQDN name defined in the OCS. **Note:** Occasionally this alarm may be activated without real cause. Resetting the MCU may clear the alarm.

Troubleshooting

- At the end of the installation and configuration process, to test the solution and the integration with the OCS, create an ongoing conference with two participants, one dial-in and one dial-out and connect them to the conference.
- If the *active* Alarm "*SIP TLS: Registration server not responding*" is displayed, stop and restart the OCS Pool Service.
- If the communication between the OCS and the RMX cannot be established, one of the possible causes can be that the RMX FQDN name is defined differently in the DNS, OCS and RMX. The name must be defined identically in all three devices, and defined as type A in the DNS. The definition of the RMX FQDN name in the DNS can be tested by pinging it and receiving the RMX signaling IP from the DNS in return.
- The communication between the OCS and the RMX can be checked in the Logger files:
  - SIP 401/407 reject messages indicate that the RMX is not configured as Trusted in the OCS and must be configured accordingly.
  - SIP 404 reject indication indicates that the connection to the OCS was established successfully.

Known Issues

- Selecting **Pause my Video** in OC client causes the call to downgrade to audio only call if the call was not in Audio Only mode at all (the call was started as a video call).

  If the call is started as an audio only call and video is added to it, or if the call was started as video call and during the call it was changed to Audio Only and back to video, selecting *Pause my Video* will suspend it as required.
- Rarely, the OC client disconnects after 15 minutes. The OC client can be reconnected using the same dialing method in which they were previously connected (dial-in or dial-out).
- Rarely, all SIP endpoints disconnect at the same time. The SIP endpoint can be reconnected using the same dialing method in which they were previously connected (dial-in or dial-out).